Lucent Technologies
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5ESS® Switch

Session Initiation Protocol (SIP) - OA&M Manual

5E16.2 and Later Software Releases

235-200-118
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About this information product

Purpose

The *5ESS® Switch Session Initiation Protocol (SIP) - OA&M Manual, 5E16.2 and Later,* 235-200-118, information product enables users, planners, maintenance personnel, engineers, installers, administrators, and provisioners to perform the necessary tasks required to support configuration, installation, monitoring, and repair of SIP signaling for packet trunking (including packet groups, SIP, SCTP, UDP, IP, QLPS connectivity, processor groups, and signaling-related hardware). OA&M procedures related to the OIU-IP bearer for SIP-signaled calls are covered in *Optical Interface Unit - Internet Protocol (OIU-IP) Interface Specification [5E16.2 and Later Software Releases],* 235-900-316.

This information product should be used as the source of complete details on this protocol to clarify the implementation on the switch and the interpretation of technical reference (TR) requirements. These details describe the *5ESS®* switch offerings in terms of the support of Session Initiation Protocol (SIP). SIP is an evolving platform in which new features will be introduced continuously for new revenue opportunities, for improved operational efficiencies, and for support of specific applications. Beginning with the 5E16.2 software release, the

5ESS® switch supports the protocols and services defined by *Telcordia Technologies, Inc.*®

This information product is expected to change as requirements and standards evolve. Therefore, Lucent Technologies reserves the right to change or delete any portion of the document, or to add information in future issues.

Reason for reissue

This information product is being reissued to support feature 99-5E-8645, "DRM IP Trunking"

Safety labels

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CAUTION

Caution indicates the presence of a hazard that can or will cause minor injury or property damage.

WARNING

Warning indicates the presence of a hazard that can cause death or severe injury.

DANGER

Danger indicates the presence of a hazard that will cause death or severe injury.

Intended audience

The *5ESS*® Switch Session Initiation Protocol for Packet Trunking - OA&M Manual describes the architecture, engineering, provisioning, and maintenance of SIP on the *5ESS*® switch. It is published as a guide for the users, planners, maintenance personnel, engineers, installers, administrators, and provisioners to perform their SIP-related tasks. Responsible personnel should have a working knowledge of telephony, switching, routing, and networking technologies.

How to use this information product

Each chapter in this information product groups related information about the SIP application.

Chapter	Content
1. Overview	Overview of SIP from an industry perspective and Lucent Technologies' perspective.
2. Architecture	Overview of the SIP architecture, starting at the Network view, then narrowing down to the System view, and finally, focusing on the hardware view.
3. Call Flow	Overview of SIP call flows through the network (office-to-office), and through the *5ESS*® switch hardware.
4. Engineering Considerations	Things to keep in mind when engineering SIP into your switch and network (for example, capacity, configuration, constraints, IP addressing schemes, network management, performance monitoring, and OS impact).
5. Provisioning	Procedures required for provisioning all aspects of the SIP platform on your switch and in your network.
6. Deprovisioning	Procedures required to remove the SIP application from the *5ESS*® switch.
7. Maintenance Considerations	Overview of troubleshooting and maintenance specific to SIP.
8. Glossary	List of acronyms used within this IP and their expansions.
9. Index	Index of subjects covered within this IP.

Conventions used

No special or unusual conventions are used in this information product.

Systems supported

This information product supports software releases 5E16.2, Feature Release 3, and later.

Related documentation

Other customer documentation that will be useful in understanding the *5ESS*® switch SIP platform are the following:

- OIU-IP feature description, *Feature Descriptions*, 235-190-400
- *OIU-IP Interface Specification*, 235-900-316
- *Session Initiation Protocol (SIP) - Interface Specification*, 235-900-344

Specific procedural and descriptive information from the following information products are referred to:

- *Administration and Engineering Guidelines*, 235-070-100
- *Translation Guide (TG-5)*, 235-080-100
- *Hardware Description*, 235-100-200
- *System Maintenance Requirements and Tools*, 235-105-110
- *Routine Operations and Maintenance Procedures*, 235-105-210
- *Corrective Maintenance Procedures*, 235-105-220
- *Hardware Change - Growth Procedures*, 235-105-231
- *System Recovery*, 235-105-250
- *Hardware Change - Degrowth Procedures*, 235-105-331
- *Recent Change Reference 5E16.2 Software Release*, 235-118-258
- *Translations Data Reference*, 235-600-126
- *Translations & Dynamic Data Reference*, 235-600-228
- *Dynamic Data Reference*, 235-600-237
- *Translations & Dynamic Data Domain Description*, 235-600-248
- *Audits Manual*, 235-600-400
- *Asserts Manual*, 235-600-500
- *Input Messages Manuals*, 235-600-700
- *Output Messages Manuals*, 235-600-750

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1 Overview

Overview

Purpose

The purpose of this chapter is to provide an overview of the following Session Initiation Protocol (SIP) features on the 5E-XC and 5E DRM:

- "SIP for Packet Trunking" (1-2) (99-5E-8382),
- "UDP Transport Layer for SIP" (1-3) (99-5E-8581),
- "Support for SIP without Preconditions" (1-3) (99-5E-8582),
- "SIP Support for Line to Packet Trunk Calls" (1-3) (99-5E-8583),
- "SIP without Encapsulated ISUP" (1-3) (99-5E-8587), and
- "SIP Enhancements to Support PSTN Gateway Phase 1" (1-3) (99-5E-8658).
- "DRM IP Trunking" (1-4) (99-5E-8645),

☐

SIP for Packet Trunking

Platform definition

Session Initiation Protocol (SIP) is a method of establishing, maintaining, and terminating internet sessions. These sessions interactively exchange real-time multimedia data (voice, text, and video) between multiple participants. SIP is based on a client-server model with intelligent end-points. End servers respond to session initiation requests and locate the called parties. SIP is an application layer protocol that can run on top of different transport protocols.

Feature descriptions

The SIP for Packet Trunking - NAR (99-5E-8382) feature adds a SIP packet trunking interface to the *5ESS*® switch. This feature allows the *5ESS*® switch to operate as a gateway for interworking between the time division multiplexing (TDM)-based public switched telephone network (PSTN) trunks and the next-generation internet protocol (IP) network virtual trunks. It interworks current circuit signaling protocols, such as Signaling System 7 (SS7) integrated services user part (ISUP), multifrequency (MF), or dual tone multifrequency (DTMF), with SIP trunk signaling over IP networks.

The SIP signaling protocol for telephony is the packet signaling protocol used to establish calls using the Optical Interface Unit - Internet Protocol (OIU-IP).

The SIP for Packet Trunking feature contains:

- two new PSU2 protocol handlers (SIP-PH and the GQPH) for setting up the SIP signaling connection,
- a new lower layer transport protocol, Stream Control Transmission Protocol (SCTP), and
- integrated operations, administration, maintenance and provisioning (OAM&P) using the conventional *5ESS*® switch interfaces.

Automatic ACM Timer - is a modification to the 99-5E-8382 feature being implemented in software release 5E16.2 FR6. The Automatic address complete message (ACM) timer is a new field on RC/V 5.82. The timer value ranges from 4-14 seconds, with 14 seconds as the default. This timer is started at the originating packet switch (OPS) by the SIP terminal process when an INVITE message is sent to a terminating packet switch (TPS). If the timer expires, the SIP Terminal Process generates a default ISUP ACM that is sent to the

ISUP originating terminal process (OTP). The ISUP OTP will send an ACM message out to the ISUP network. This prevents the call from being terminated by the ISUP office because the TPS response was too slow.

The features that follow enhance the capabilities provided by the SIP for Packet Trunking feature (99-5E-8382). Feature descriptions for these features can be found in the *Feature Descriptions*, 235-190-400 document.

UDP Transport Layer for SIP

The UDP Transport Layer for SIP (99-5E-8581) feature allows the *5E-XC*™ to support SIP with a UDP transport layer instead of SCTP, allowing the *5E-XC*™ to connect to SIP network elements that do not support SCTP.

Support for SIP without Preconditions

The Support for SIP without Preconditions (99-5E-8582) feature allows SIP calls to be established without the precondition procedures that were initially required by the SIP for Packet Trunking feature. For additional information about SIP signaling procedures, with or without preconditions, see the *Session Initiation Protocol (SIP) - Interface Specification*, 235-900-344 document.

SIP Support for Line to Packet Trunk Calls

The SIP Support for Line to Packet Trunk Calls (99-5E-8583) feature allows analog and ISDN line originations and terminations to be routed to and from SIP packet groups, in addition to the circuit trunk originations and terminations initially supported by the SIP for Packet Trunking feature.

SIP without Encapsulated ISUP

The SIP without Encapsulated ISUP (99-5E-8587) feature allows the *5E-XC*™ to interwork with elements that do not generate and process ISUP messages and to generate appropriate interworking messages to pass through the network. The *5E-XC*™ supports SIP without Encapsulated ISUP on a per packet group basis.

SIP Enhancements to Support PSTN Gateway Phase 1

The SIP Enhancements to Support PSTN Gateway Phase 1 (99-5E-8658) feature provides a series of enhancements that allow phone calls to be setup between PSTN subscribers and IP subscribers that terminate to a Telephone Application Server (TAS). The PSTN

Gateway sits between the IP network (using SIP signaling) and PSTN network (using ISUP signaling). The PSTN Gateway performs the interworking between SIP and ISUP protocols.

DRM IP Trunking

The DRM IP Trunking (99-5E-8645) feature extends SIP-T signaling and OIU-IP packet trunking to the DRM.

Feature descriptions for the features listed above can be found in the *Feature Descriptions*, 235-190-400 document.

Benefits

SIP is an emerging IP-based protocol that is critical for deploying converged and next generation real-time voice, data and video communication services. SIP for Packet Trunking gives the *5ESS®* switch direct access to more efficient IP transport networks. It allows for the utilization of more cost-effective revenue-generating services. It also can be used to expand trunking capacity or to replace existing circuit trunks.

SIP for Packet Trunking:

- reuses the embedded network equipment to migrate to an all IP multimedia network,
- provides high quality service with 99.999% hardware and software reliability,
- reuses existing integrated *5ESS®* switch OAM&P interfaces,
- supports inter-operability with other vendors equipment through development based on industry standards,
- reuses existing switch hardware and software infrastructure,
- interfaces with the TDM-based PSTN trunks or next generation IP network virtual trunks, and
- can use SCTP to provide additional network security.

Differences

The existing TDM network contains dedicated circuits between two end-points. IP calls have no fixed connections. Calls are dynamically routed through the packet network based on available bandwidth. The connection information is exchanged using SIP signaling.

The migration from an expansive SS7-based signaling transfer point (STP) network to an IP-based signaling network will be ongoing. As SIP evolves, more capabilities will be defined and added.

Secured features

The features that follow are secured features listed with the associated secured feature IDs (SFIDs). Recent Change view 8.22 is used to activate these features. Refer to "Feature Activation (RC/V 8.22)" (5-13) for the procedure.

- SIP for Packet Trunking (SFID 684)
- SIP without Encapsulated ISUP (SFID 769)

Availability

The following list provides the availability for SIP related features:

- SIP for Packet Trunking (feature 99-5E-8382) is available with the 5E16.2 FR3 software release or later.
- UDP Transport Layer for SIP (feature 99-5E-8581) and Support for SIP without Preconditions (feature 99-5E-8582) are available with 5E16.2 FR5 software release or later.
- SIP without Encapsulated ISUP (feature 99-5E-8587) and SIP Enhancements to Support PSTN Gateway Phase 1 (feature 99-5E-8658) are available with the 5E16.2 FR6 software release or later.
- The DRM IP Trunking (99-5E-8645) feature is available with the 5E16.2 FR 9 software release or later.

Deployment

This platform is provided on a per office basis.

Background knowledge

When SIP is used in media gateway controllers for call establishment, there are many cases where it must bridge the PSTN network with IP networks. To do this, SIP must be extended to transport all of the PSTN signaling information transparently. By extending SIP messaging and adding PSTN signaling encapsulation functionality, the SIP protocol can be used for media gateway to media gateway communication.

Interfacing with a IP network requires an understanding of the following IP concepts and equipment:

- routers and Layer 2 (L2) switches,
- Ethernet,
- internet protocol and internet control message protocol (ICMP),

- transport protocols like SCTP and user datagram protocol (UDP), and
- SIP standards and protocols.

Hardware Dependencies

The Optical Interface Unit - Internet Protocol (OIU-IP) hardware (feature 99-5E-8308) needs to be installed and active.

Other hardware required to support this feature includes two types of protocol handlers (PHs); the PHE2 to support the SIP PH functionality, and the PH33 to support the General Quad-link Package Switch (QLPS) PH (GQPH), as well as the following:

- SM2000 switching modules,
- QLPS,
- Packet Switch Unit - Model 2 (PSU2), and
- Data Fanout - Model 2 (DF2).

Note: This QLPS and PH33 are not required on the DRM/VCDX.

Refer to Chapter 4, “Engineering Considerations” for a list of the required hardware.

Software Dependencies

The following software components are needed:

- new SIP signaling software supported by SCTP or UDP for transport,
- software to support the optical facility interface (OFI) circuit pack hardware delivered with the Optical Interface Unit for NAR Market feature (99-5E-7140),
- software to support the optical facility interface internet protocol (OFI-IP) circuit pack hardware delivered with the Optical Interface Unit - Internet Protocol feature (99-5E-8308), and
- operational support system (OSS) enhancements.

Incompatibilities

There are no incompatibilities with other features.

Feature Interactions

SIP for Packet trunking defines the signaling protocol to control the bearer transport delivered with the Optical Interface Unit - Internet Protocol (OIU-IP) feature (99-5E-8308).

SIP for Packet Trunking that uses UDP Transport Layer for SIP (99-5E-8581) requires Support for SIP without Preconditions

(99-5E-8582). Both features are dependent on the base SIP feature (99-5E-8382).

The SIP without Encapsulated ISUP feature (99-5E-8587) is dependent on the base SIP feature (99-5E-8382). The SIP without Encapsulated ISUP feature is transport independent.

The SIP Enhancements to Support PSTN Gateway Phase 1 (99-5E-8658) is dependent on the base SIP feature (99-5E-8382).

The DRM IP Trunking (99-5E-8645) is dependent on the base SIP feature (99-5E-8382).

Inter-operability

One of the main objectives of SIP is to provide call signaling and call control independent of the IP network technology.

Inter-operability with IP equipment

The IP router or Layer 2 (L2) switch to which the *5ESS®* switch protocol handlers (PHs) interface can be from any vendor, provided that it meets the requirements of the *Session Initiation Protocol (SIP) - Interface Specification,* 235-900-344 document.

The IP network components and the *5ESS®* switch can be configured many different ways. The different configurations are described in "IP Router & Layer 2 Switch Interoperability" (2-15).

Inter-operability with other Switching Systems

SIP for Packet Trunking is based on standards for the protocol layers including SIP, SCTP, and/or UDP, IP, and ICMP, and the software is intended to operate with other vendors. However, this feature works best when used with other *5ESS®* switches. If another vendor's switch adheres to the standards in the same way, the products should be compatible. No assumptions are made about inter-operability with other vendor's equipment.

The *Session Initiation Protocol (SIP) - Interface Specification,* 235-900-344 document explains the interface implementation in greater detail.

Customer Security

It is the customer's responsibility to ensure the security of the IP backbone network. It is assumed that the IP layer security is performed by the edge router and that the *5ESS®* switch is in a trusted network.

Industry standards

SIP for Packet Trunking is based on standards by the Internet Engineering Task Force (IETF) and *Telecordia*™ Technologies.

SIP standards continue to be defined and extended. Some standards are in draft form and, in some cases, incomplete or not well defined. In some instances, standards could not be followed simply because no standard exists.

IETF RFC	Title
RFC 3261	SIP: Session Initiation Protocol
RFC 2960	Stream Control Transmission Protocol
RFC 768	User Datagram Protocol
RFC 791	Internet Protocol
RFC 792	Internet Control Message Protocol
RFC 1122	Requirements for Internet Hosts -- Communication Layers
RFC 1332	The PPP Internet Protocol Control Protocol (IPCP)
RFC 1661	The Point-to-Point Protocol (PPP)
RFC 1662	PPP in HDLC-like Framing
RFC 2474	Definition of the Differentiated Services Field (DS Field) in the IPv4 and IPv6 Headers
RFC 2615	PPP over SONET/SDH
RFC 1878	Variable Length Subnet Table for IPv4
RFC 1918	Address Allocation for Private Internets
RFC 917	Internet Subnets
RFC 1519	Classless Inter-Domain Routing (CIDR): an Address Assignment and Aggregation Strategy
RFC 2616	Hypertext Transfer Protocol -- HTTP/1.1
RFC 2665	Definitions of Managed Objects for the Ethernet-like Interface Types
RFC 2011	SNMPv2 Management Information Base for the Internet Protocol using SMIv2
RFC 2013	SNMPv2 Management Information Base for the User Datagram Protocol using SMIv2
RFC 3372	Session Initiation Protocol for Telephones (SIP-T): Context and Architectures

IETF RFC	Title
RFC 3204	MIME Media Types for ISUP and QSIG Objects
RFC 2327	SDP: Session Description Protocol
DRAFT RFC 3515	The Session Initiation Protocol (SIP) Refer Method
DRAFT	Redirection Service Capability in the Session Initiation Protocol (SIP)
DRAFT	The Stream Control Transmission Protocol as a Transport for the Session Initiation Protocol
RFC 3312	Integration of Resource Management and Session Initiation Protocol (SIP)
RFC 3262	Reliability of Provisional Responses in the Session Initiation Protocol (SIP)
DRAFT	Session Initiation Protocol Service Examples
DRAFT	Using ENUM for SIP Applications
RFC 3398	ISUP to SIP Mapping

Telcordia Technologies GRs	Title
GR-253-CORE	Synchronous Optical Network (SONET) Transport Systems: Common Generic Criteria
GR-474-CORE	Network Maintenance: Alarm and Control for Network Elements
GR-782-CORE	SONET Digital Switch Trunk Interface Criteria
GR-3053-CORE	Voice over Packet (VoP): Next Generation Network (NGN) Signaling Gateway Generic Requirements

2 Architecture

Overview

Purpose

This chapter contains information about the Session Initiation Protocol (SIP) architecture and how it is integrated within the *5ESS*® switch.

This chapter describes the SIP architecture from a:

- network view,
- system view,
- signaling view,
- hardware view, and
- security view.

Network View

The network view describes the SIP architecture at a network level between network elements.

System View

The system view contains SIP architecture information within a *5ESS*® switch.

Signaling View

The signaling view explains the following SIP signaling transports

- Stream Control Transmission Protocol (SCTP)
- User Datagram Protocol (UDP)

Hardware View

The hardware view provides information about the various hardware components that are required to implement SIP for Packet Trunking.

Security View

The security view describes the cookie mechanism implemented in SCTP and how it protects the *5ESS*® switch.

□

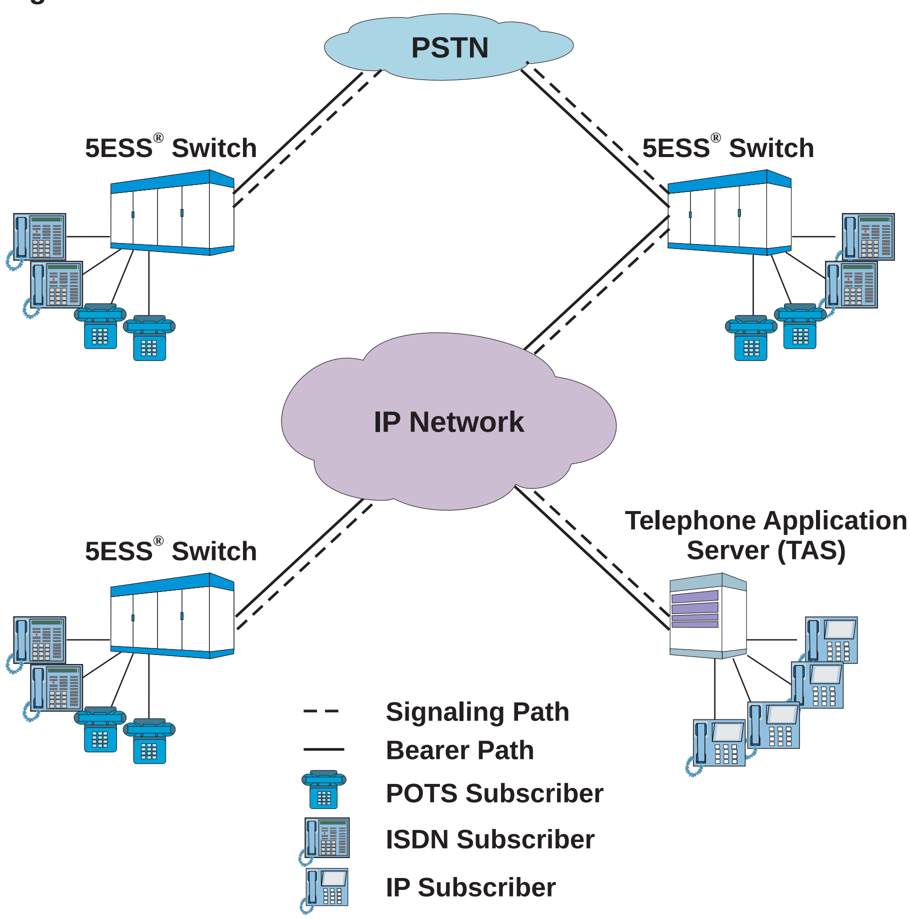
Network View

Overview

The *5ESS®* switch interfaces to internet protocol (IP) packet networks using integrated packet trunking. SIP is a packet call signaling protocol that offers a means to inter-operate with packet trunking interfaces between network elements. SIP signaling uses the packet switch unit model 2 (PSU2) to establish calls on the optical interface unit-internet protocol (OIU-IP) integrated gateway.

IP Backbone

Figure 2-1 IP Backbone

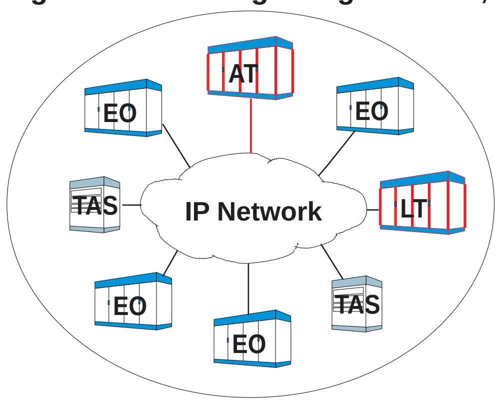


SIP for packet trunking and its related features provide the functionality that allows the *5ESS®* switch to interface to an IP network. The signaling network, which uses session initiation protocol (SIP), and bearer network can be located on distinct physical networks, on one common network, or on distinct logical networks over a single physical network. Figure 2-1, "IP Backbone" (2-3) illustrates one common network. Figure 2-5, "Packet Trunking" (2-7) illustrates distinct networks.

Signaling Network

From a hardware perspective, each network element, such as end offices (EO), EO-local tandems (LT), and EO-access tandems (AT), in the SIP signaling network is connected to the IP network. Signaling messages between network element are dynamically routed by the IP network.

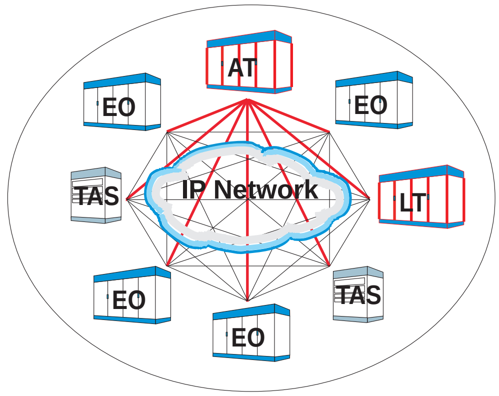
Figure 2-2 SIP Signaling Network, Hardware Perspective



EO = End Office
AT = Access Tandem
LT = Local Tandem
TAS = Telephone Application Server

Figure 2-2, "SIP Signaling Network, Hardware Perspective" (2-4) shows the signaling network for an IP network using SIP from a hardware perspective. The IP network fully interconnects all of the network elements.

Figure 2-3 SIP Signaling Network, Provisioning Perspective



EO = End Office
AT = Access Tandem
LT = Local Tandem
TAS = Telephone Application Server

Figure 2-3, "SIP Signaling Network, Provisioning Perspective" (2-5) shows the signaling network from a provisioning perspective. Each network element must know about all other network elements to which it is interconnected.

Bearer Network The bearer network looks the same from both a hardware and provisioning perspective. Each termination is defined but there are no fixed paths. The path between two network elements is allocated dynamically on a call by call basis.

Figure 2-4 Bearer Network

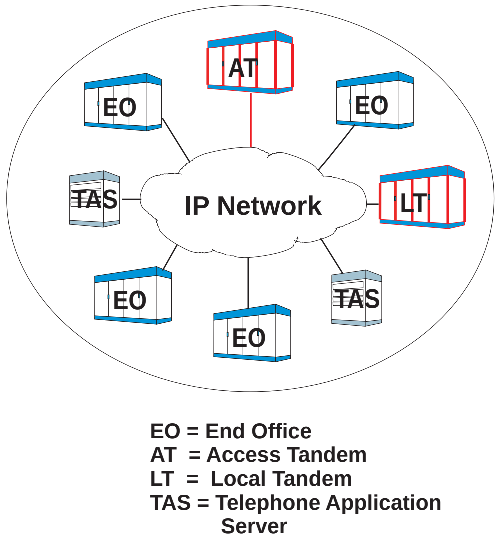


Figure 2-4, "Bearer Network" (2-6) shows that the bearer trunks are connected to the central IP hub. For a packet call between two network elements, an IP association is created between two endpoints, but no fixed paths through the IP network are provisioned.

Packet Trunking

Figure 2-5 Packet Trunking

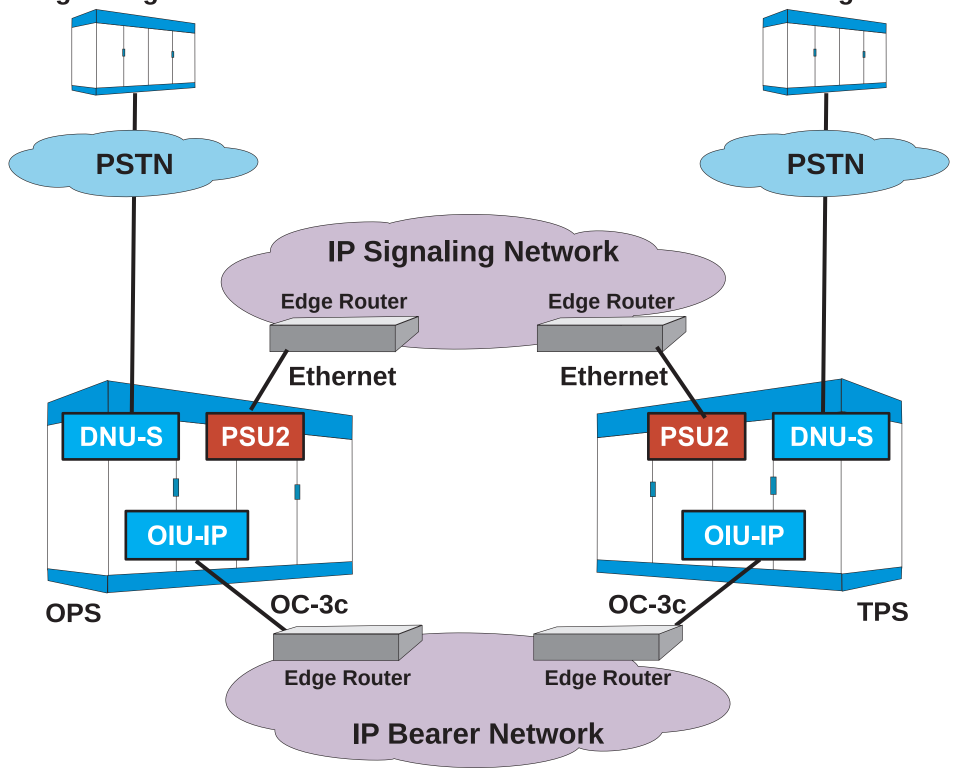
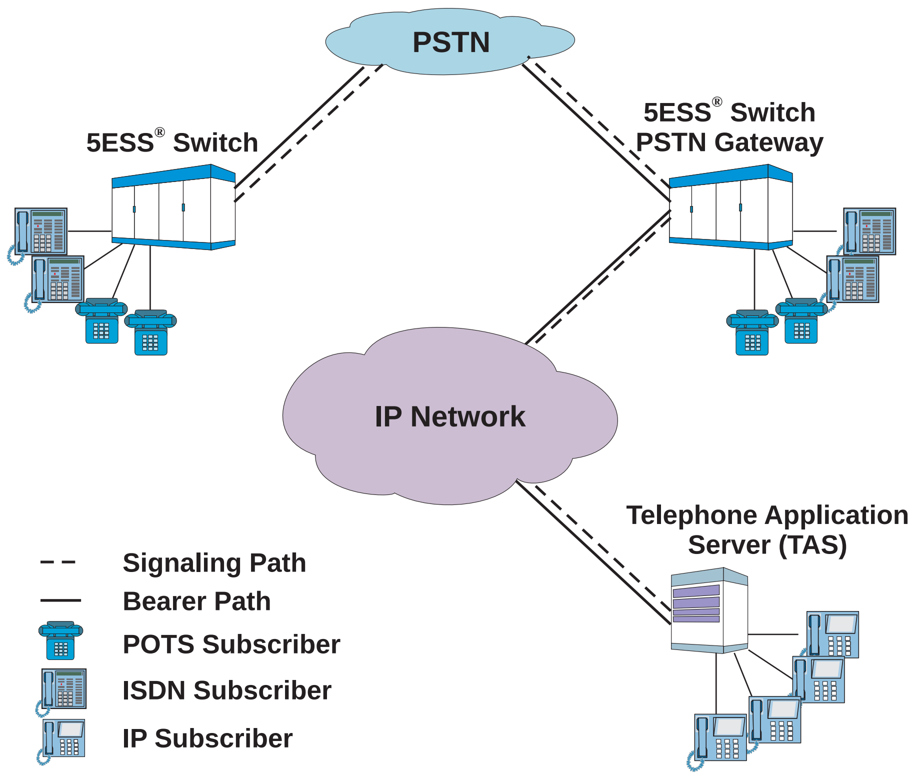


Figure 2-5, "Packet Trunking" (2-7) illustrates a packet trunking network. In this network, *5ESS*® switches connect to IP switches and IP routers connected in order to provide end-to-end routing of information. An end office (EO), the *5ESS*® switch, connects to an edge router to access the IP network. The two packet offices shown are *5ESS*® switches; however, either can be other vendor switches as long as they support IP packet trunking with SIP signaling. An Ethernet link connects the PSU2 to an edge router to provide the signaling path. A Synchronous Optical Network (SONET) optical carrier - level 3 concatenated (OC-3c) link connects the OIU-IP and the router for the voice path.

Calls using packet trunking are dynamically allocated to available packet network interface bandwidth each time a call is established, whereas calls using time division multiplexing (TDM) trunking are assigned to circuits or connections dedicated to two endpoints. In packet trunking, call and connection information is exchanged using SIP signaling and voice packets are transmitted, routed and received between two switches using the pair of dynamically allocated packet interfaces that acts as a virtual trunk.

Public Switched Telephone Network (PSTN) Gateway

Figure 2-6 PSTN Gateway



The PSTN Gateway application allows phone calls to be setup between PSTN subscribers that terminate to a *5ESS*® switch and IP subscribers that terminate to a Telephone Application Server (TAS). The PSTN Gateway sits between the IP network (using SIP signaling) and PSTN network (using ISUP signaling). Figure 2-6, “PSTN Gateway” (2-8) illustrates how the PSTN gateway fits into the network. The PSTN Gateway performs the interworking between SIP and ISUP protocols.

□

System View

Overview This section identifies the different components and describes their function in the SIP architecture.

Architecture The SIP for Packet Trunking feature advances the architecture of the *5ESS*® switch with the introduction of several components. Existing hardware and software components, including the administrative module (AM), communication module (CM), switching module 2000 (SM-2000), and peripheral units, are preserved.

Figure 2-7 SIP Architecture

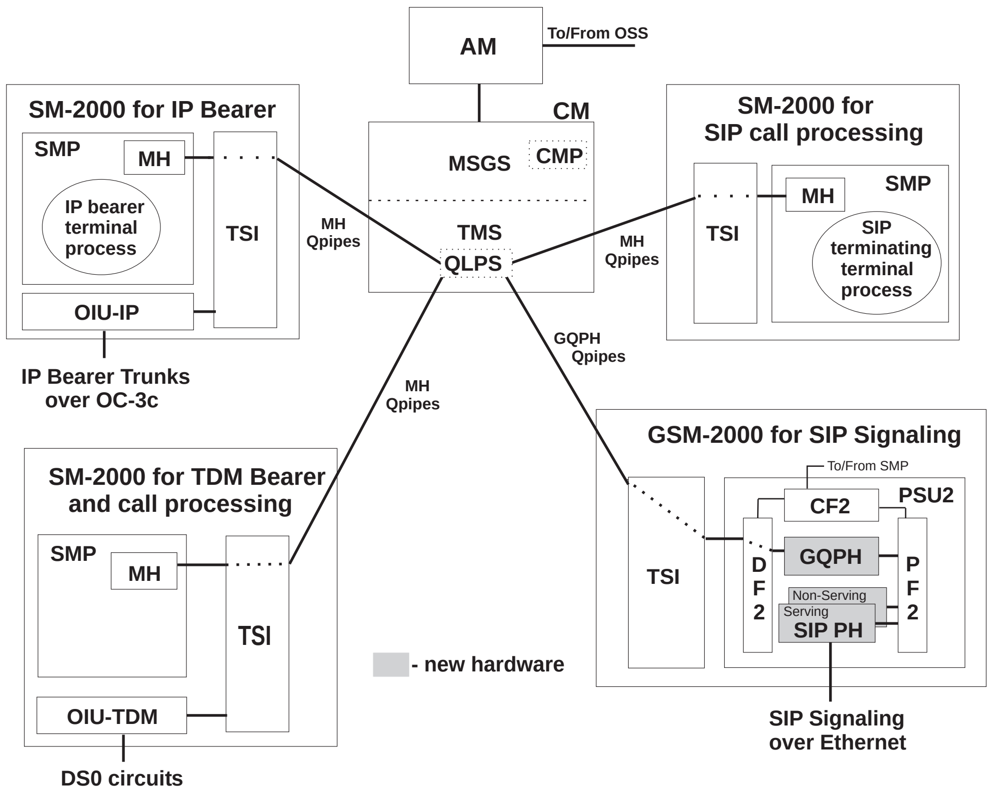


Figure 2-7, “SIP Architecture” (2-9) illustrates the SIP call signaling and bearer architecture.

The signaling components within the packet switch unit model 2 (PSU2) include the:

- general QLPS protocol handler (GQPH), and
- session initiation protocol - protocol handler (SIP PH).

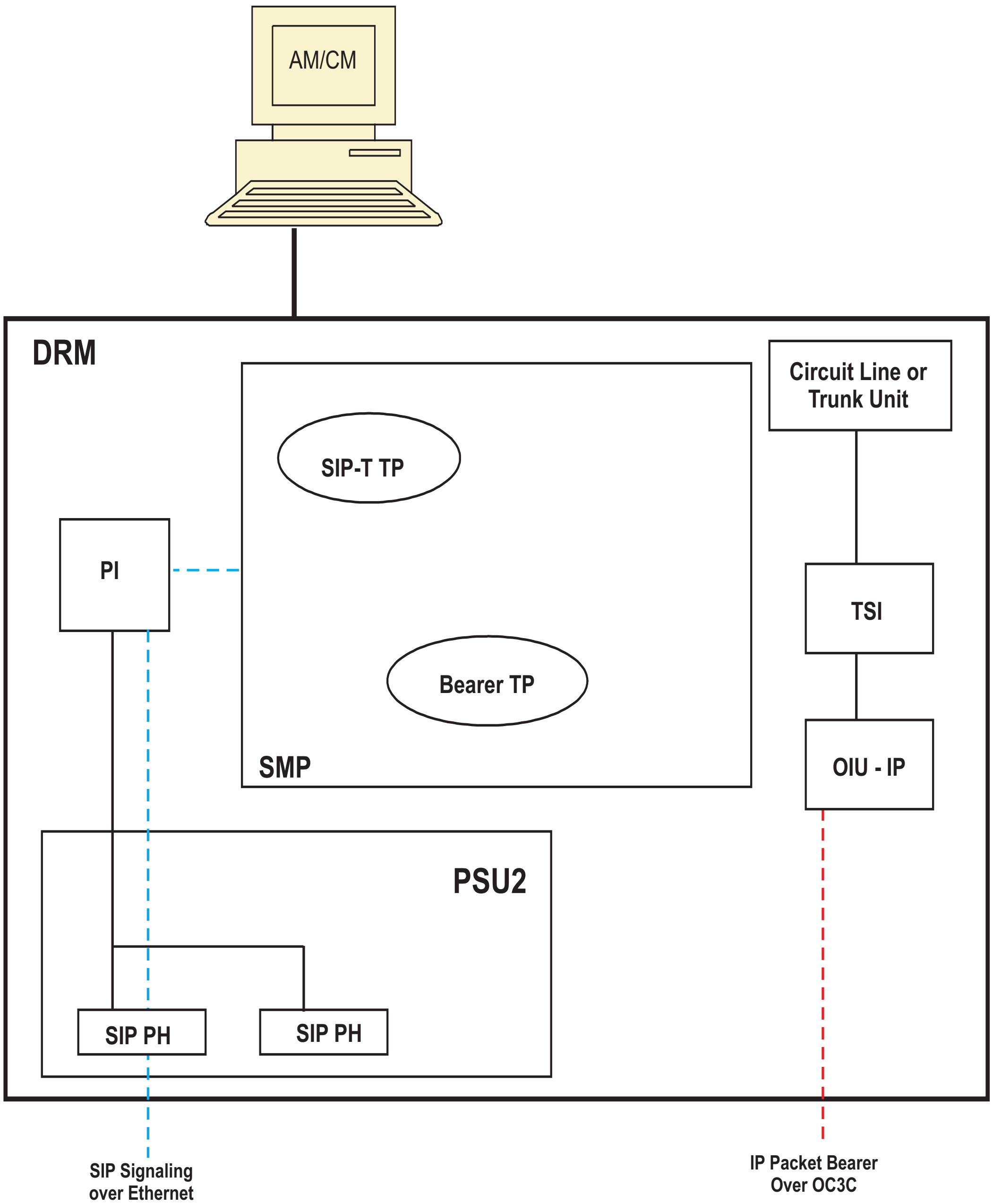
Note: The GQPH is only required on for non-DRM/VCDX.

The packet trunking bearer is handled by the optical interface unit - internet protocol (OIU-IP).

Refer to Figure 2-7, “SIP Architecture” (2-9) when reading the following hardware descriptions.

Note:Figure Figure 2-7, “SIP Architecture” (2-9) is the non-DRM/VCDX architecture. See figure Figure 2-8, “SIP on DRM Architecture” (2-11) showing the architecture on the DRM .

Figure 2-8 SIP on DRM Architecture



Administrative Module (AM)

SIP operations, administration, maintenance and provisioning (OAM&P) are integrated within the *5ESS*® switch. The administrative module (AM) is the integrated element management system (EMS) of the *5ESS*® switch and provides the interface to the operations support systems (OSSs). The OSS network includes the circuit OSS products for traditional public switched telephone network (PSTN) equipment.

Communication Module (CM)

The CM hardware remains unchanged. The CM consists of the message switch (MSGS) and the time multiplexed switch (TMS).

The communication module processor (CMP) is one of several functions within the MSGS. To support the SIP feature, the CMP selects an SM-2000 to process SIP call signaling for outgoing calls when destined for routes over packet groups controlled by SIP signaling. The CMP also selects the SM-2000 with an available OIU-IP for the bearer connection for both incoming and outgoing calls. ***Note:*** The CMP functions for SIP on the non-DRM/VCDX described here are carried out on the Switching Module Processor (SMP) for SIP on the DRM (other than selecting an SM, which isn't required on DRM, since there is only the one SM).

Within the TMS, the quad link packet switch (QLPS) terminates QLPS endpoints, e.g. message handlers (MHs). SIP introduces the GQPH which is a new QLPS endpoint. Messages are sent between the QLPS and GQPH over a logical connection called a GQPH Qpipe. Connections between the GQPH and the call processing SMs are established over QLPS logical links. These connections are used to carry SIP related signaling messages between the SMs and the SIP PHs. The fabric of the TMS is used when different SMs are used for the TDM and IP bearer connections. In addition, any messaging between two SMs and between an SM and the AM goes through the CM. ***Note:*** This paragraph only applies to non-DRM/VCDX, there is no QLPS or GQPHs on the DRM/VCDX.

Global Switching Module-2000 (GSM-2000)

The GSM-2000 supports the PSU2. It is recommended that this module be set-up to provide only signaling.

Below is a partial list of status tracked by the GSM-2000 when supporting the SIP feature:

- SIP PHs,
- Ethernet links,
- processor groups,

- GQPHs, and
- SCTP associations and association sets.

Note: The GQPHs are only on the non-DRM/VCDX.

Packet Switch Unit Model 2 (PSU2)

The PSU2 is the signaling message distribution point for SIP messages between the near *5ESS*® switch and the far *5ESS*® switch. To support the SIP feature the PSU2 is equipped with SIP PHs and GQPHs.***Note:*** The GQPHs are only on the non-DRM/VCDX.

Session Initiation Protocol Protocol Handler (SIP PH)

The SIP PH is a logical protocol handler that uses the PHE2 hardware. The SIP PH performs the message parser, message reformer, and transaction manager functions for SIP messages.

For outgoing messages, the SIP signaling terminal process in a SMP sends SIP related call information to the SIP PH and the SIP PH constructs a:

1. SIP header message with call routing information, and a
2. SIP message body with application data.

For incoming messages, the SIP PH:

1. validates and translates SIP messages, and
2. delivers the data to the SIP signaling terminal process for processing by the call processing programs.

The SIP PH converts message data between SIP messages and an internal protocol used by the SMP. The internal messages are transported between the SIP PH and the SMP by the GQPH.

The SIP PHs are configured as processor groups each having one or two PHs. When redundancy is desired and processor groups are provisioned in pairs, one SIP PH will dynamically be assigned the “serving” role and the other as the “non-serving” role. During normal operations, the serving SIP PH provides the non-serving SIP PH with stable call data. Should the serving SIP PH fail, stable calls will be preserved. Refer to “Serving/Non-Serving” (2-30) for more details.

A 100BaseT Ethernet interface is used to pass SIP messages between the serving SIP PH and the IP network.

General QLPS Protocol Handler (GQPH)

The GQPH is a logical protocol handler that uses the PH33 hardware type. The GQPH supports message routing between a SIP PH and the SMP used for SIP call control. For outgoing messages sent from the SMP to the SIP PH, the GQPH forwards messages to the serving PH. For incoming calls, the GQPH forwards messages from the SIP PH to one of the QLPSs, which then forwards the message to the correct SM. The GQPH communicates with the QLPS network over logical connections called GQPH Qpipes.***Note:*** The GQPHs are only on the non-DRM/VCDX.

Switching Module-2000 (SM-2000)

The SM-2000 provides IP packet trunking. The SMP terminal processes coordinates establishing the signaling and bearer connections between the originating and terminating network element. The bearer terminal process selects the OFI-IP for the voice connection. The signaling terminal process coordinates call information between the two network elements.

Optical Interface Unit-Internet Protocol (OIU-IP)

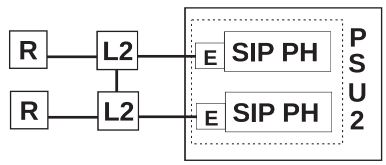
The OIU-IP on the *5ESS*® switch provides the packet network interface for bearer traffic. Calls are switched in the time slot interchange (TSI) and an optical facility interface-internet protocol (OFI-IP) in the OIU-IP performs synchronous-to-asynchronous conversion (SAC) to transport the voice stream from the TDM network to the packet network.

The OFI-IP hardware control software interacts with the SM-2000 switching module processor (SMP) peripheral control using the proprietary protocol. Real-time maintenance orders and reports are also sent using the proprietary protocol to support OAM&P activities. An OFI-IP protection group (PG) supports interconnection to an IP edge router using OC-3c facilities with 1+1 automatic protection switching (APS).

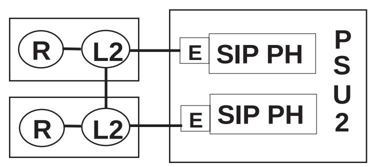
IP Router & Layer 2 Switch Interoperability

The *5ESS*® switch can interface with a number of IP network configurations. Figure 2-9, “Sample Configurations” (2-15) provides some examples of configurations that may be used.

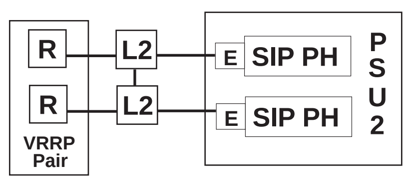
Figure 2-9 Sample Configurations



(a) Separate pairs of L2 switches and IP routers, SIP PH/Ethernet are a serving/non-serving pair.



(b) Integrated pair of L2 switches and IP routers.



(c) VRRP pair providing access routing function, plus physical L2 switches (VRRP pair appears as a single IP router to the 5ESS® switch).

Legend

R	access router
L2	Layer 2 switch
E	Ethernet link
SIP PH	SIP protocol handler
PSU2	Packet Switching Unit 2
VRRP	Virtual Router Redundancy Protocol

The configuration shown in example “a”, uses a pair of layer 2 switches along with a separate pair of routers. This configuration can be used in cases where the routers used cannot support the layer 2 functionality. When used with the SCTP capability to support multiple IP paths, as well as with the SIP PH’s ability to monitor the status of its reachability to the access routers, this configuration can achieve a significantly high level of reliability.

The configuration shown in example “b” is recommended when the router can support both the layer 2 function and routing function. This configuration would be expected to provide similar reliability to configuration “a”.

The configuration shown in example “c” using a pair of routers configured as a virtual router redundancy protocol (VRRP) pair can provide reliability in networks in where SCTP multihoming capability is not widely supported. If SCTP multihoming is widely supported, operating the routers independent of each other would be preferable.

Some considerations when selecting a configuration are the capabilities of the existing IP equipment, the desired level of reliability, the provisioning effort, and cost.

□

Signaling View

Overview

The 5E-XC supports SIP on the following transport layers:

- Stream Control Transmission Protocol (SCTP)
- User Datagram Protocol (UDP)

This section defines the concepts of endpoints, associations, and association sets for the stream control transmission protocol (SCTP), transport layer, and the concept of UDP paths for the UDP transport layer.

Figure 2-10, "SIP Protocol Stack" (2-17) shows the protocol stack supporting SIP signaling with either a SCTP or UDP transport layer.

Figure 2-10 SIP Protocol Stack

SIP
SCTP/UDP
IP/ICMP
Ethernet
100BaseT

For more detailed information about the layers of the protocol stack refer to the *Session Initiation Protocol (SIP) - Interface Specification*, 235-900-344 document.

Transport Layer Comparison

Table 2-1, "TCP/UDP/SCTP Comparison" (2-17) highlights some of the differences between the TCP, UDP, and SCTP transport layers.

Table 2-1 TCP/UDP/SCTP Comparison

TCP	UDP	SCTP
byte-oriented stream	message-oriented stream	message-oriented stream
multiple sockets use more resources	simpler protocol needs less resources	multiple streams uses less resources
single-homed	single-homed	multi-homed
vulnerable to synchronize (SYN) bit packet attacks	connectionless protocol is vulnerable to attack	4-way association startup procedure, security cookie

Stream Control Transmission Protocol (SCTP)

SIP can use stream control transmission protocol (SCTP) to provide reliable end-to-end transport over an IP based network.

The benefits of using SCTP over transmission control protocol (TCP) or user datagram protocol (UDP) for transport are:

- improvements on the weaknesses of TCP and UDP for telephony applications,
- provides the reliability of multiple paths between endpoints (per association), and
- suitability for applications beyond SIP.

The *Session Initiation Protocol (SIP) - Interface Specification,* 235-900-344 document explains the interface implementation in greater detail.

Endpoints

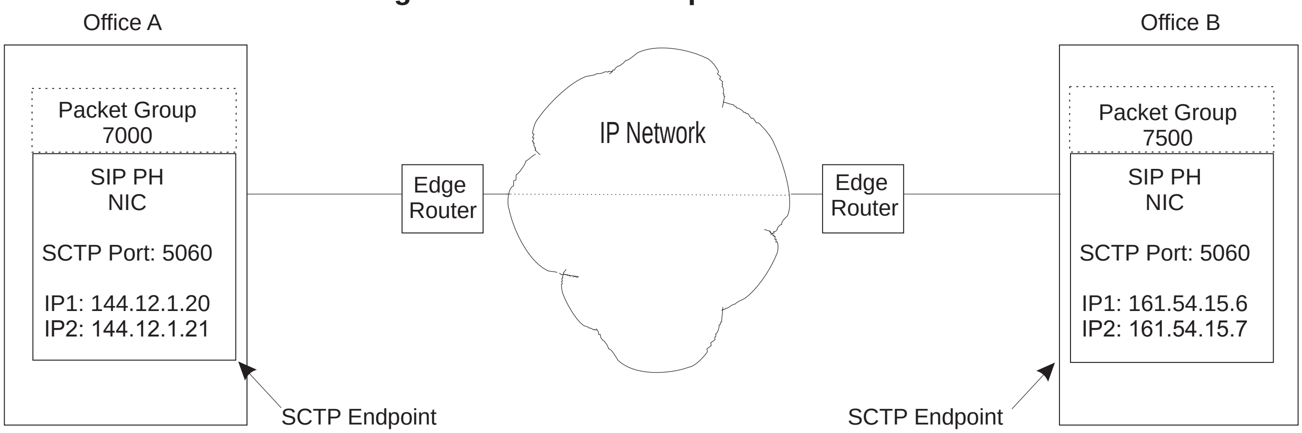
An SCTP endpoint is assigned to a processor group and each endpoint is identified by an address known as the SCTP transport address. An SCTP transport address consists of an SCTP port number and a list of one or more IP addresses. The SCTP transport address uniquely identifies an SCTP endpoint. The SCTP port number is used to send and receive messages. Each SIP PH processor group in an office can support one SCTP endpoint. Refer to Figure 2-11, "SCTP Endpoints" (2-19).

An endpoint with more than one IP address in its SCTP transport address list is called a multi-homed SCTP endpoint. An endpoint can be configured with multiple network interface cards (NIC) each with its own IP address, or multiple IP addresses can be assigned to one NIC. On the *5ESS*® switch, the SIP PH processor group is the NIC and it can be assigned two IP addresses. When two IP addresses are assigned to a SIP PH processor group, at any one time one will be the serving SIP PH and the other will be the non-serving SIP PH. Both IP addresses will be assigned to the SIP PH performing the serving function.

An upper layer protocol like SIP may need to communicate with peers physically accessed through separate IP networks. In this case, an

SCTP endpoint is established in each network on which a far, peer SIP entity resides.

Figure 2-11 SCTP Endpoints



Association

An SCTP association is a protocol relationship between SCTP endpoints. It is composed of the two SCTP endpoints and protocol state information including verification tags and the currently active set of transmission sequence numbers (TSNs). An association is uniquely identified by the transport addresses used by the endpoints in the association. Two SCTP endpoints can not have more than one SCTP association between them at any given time.

The SCTP layer initially establishes communication between two endpoints. The establishment of the SCTP association between the two endpoints is done with a cookie mechanism that results in the creation of a transmission control block (TCB). The TCB contains information about the endpoints and parameters governing how the endpoints will communicate. Figure 2-18, “Cookie Mechanism” (2-36) illustrates the exchange.

Below are the steps used to establish an association.

1. The four-way handshake is initiated by the initializing packet switch with an INIT message.

2. The far packet switch creates a cookie and sends it to the first packet switch in an ACK message.

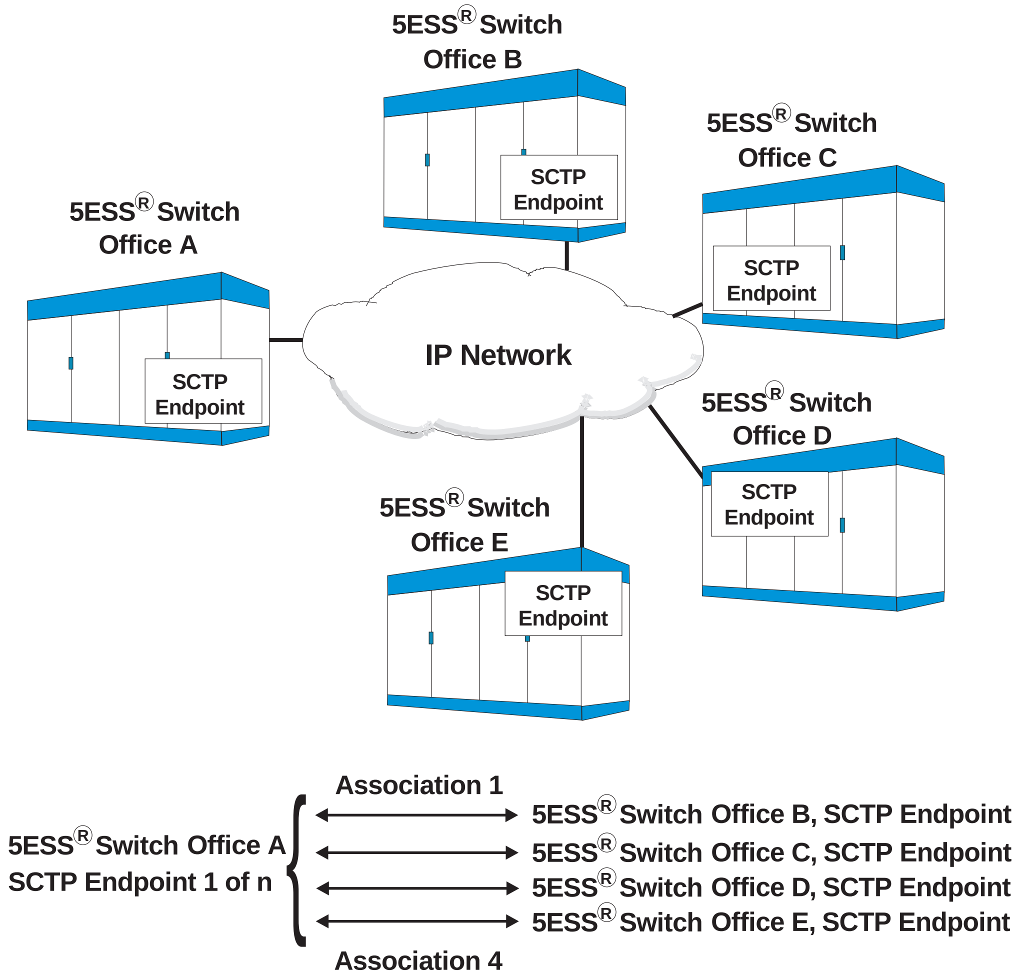
3. The first packet switch returns the cookie in an ECHO message to the far packet switch.
4. The far packet switch validates the cookie information and returns the cookie in an ACK message.

The endpoint at the initial packet switch is associated with the endpoint at the terminating packet switch through mutual knowledge by the TCBs and the implicit agreement to transmit messages to and receive messages from each other. Either endpoint in an association is allowed to initiate the establishment of the association.

Only one association may exist between two SCTP endpoints. However, a number of concurrent SCTP signaling flows may exist between those endpoints. In SCTP, these flows are called streams. Each SIP message contains an association and a stream number. Messages that are part of the same transaction are given the same stream number. For example, all SIP messages associated with a given call are carried over the same SCTP stream.

One near endpoint can also be associated with multiple far endpoints. Figure 2-12, "Multiple SCTP Associations" (2-21) illustrates office A containing one endpoint that has associations with more than one office. Although the illustration shows the office with only one endpoint, it may actually have many endpoints and associations.

Figure 2-12 Multiple SCTP Associations

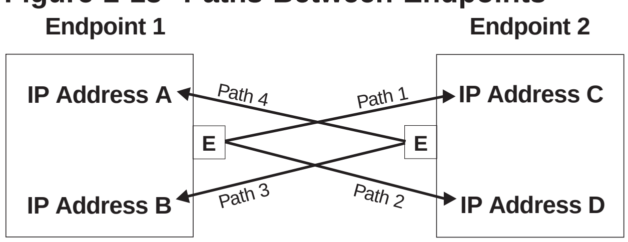


Associations in the *5ESS®* switch are static and they are determined by the provisioning. Associations are established when a SIP PH pair is initialized.

Paths

An association is a method to logically link two endpoints together to transfer data. A path is the physical route through the network between two associated endpoints. The number of paths from one endpoint to another depends on the number of transport addresses homed on each endpoint. Single-homed association endpoints have only one path between two endpoints. Multi-homed endpoints have more than one path between two endpoints. For example, if an association exists between two endpoints and both have two IP addresses assigned, there are four potential paths. Figure 2-13, "Paths Between Endpoints" (2-22) shows multi-homed endpoints (Endpoint 1 and Endpoint 2) with each SIP PH being assigned two IP addresses. The number of paths to the destination endpoint is based on the number of destination IP addresses. The number of source IP addresses is only relevant to the far packet switch. In Figure 2-13, "Paths Between Endpoints" (2-22), from Endpoint 1 to Endpoint 2 there are two paths and from Endpoint 2 to Endpoint 1 there are two paths.

Figure 2-13 Paths Between Endpoints



The advantage of multi-homed endpoints is they provide multiple paths in the IP network. If a specific path is out of service or failing, an alternate path can be used to send messages.

Association Sets

Association sets are unique to the *5ESS*® switch. An association set is a grouping of SCTP associations. The purpose of an association set for the SIP capability is to support the transport of the SIP messages used to establish calls over a particular packet group.

Association sets minimize these limitations of associations:

- capacity mismatch of the two endpoints,
- provisioning limitations for the number of associations, and
- inability to route calls on non-functioning associations.

Association set are analogous to signaling link sets in Signaling System 7 (SS7) networks.

The associations pertaining to a given association set can be assigned across multiple processor groups. Thus, the call signaling pertaining to a given packet group can be load-shared across multiple processor groups.

When routing new calls to a particular packet group, the *5ESS*® switch will avoid selecting any SCTP associations that are not functioning properly. This increases the reliability of the signaling pertaining to a packet group.

User Datagram Protocol (UDP)

SIP can use User Datagram Protocol (UDP) in place of SCTP to provide the transport layer over an IP-based network, if adjacent network elements support UDP, but not SCTP. The use of UDP transport simplifies data provisioning, but lacks some of the benefits offered by SCTP, most notably the multi-homing and multi-stream capabilities.

Because UDP is a connectionless, stateless, unreliable protocol, the SIP layer is enhanced to handle retransmission of SIP messages and to react to “Destination Unreachable” indications from the IP network when the transport layer is UDP.

Paths

The *5E-XC*™ supports provisioning of UDP paths to identify the near-end processor group, IP address and UDP port number, and far-end IP address, and UDP port number to be used by the transport layer for sending and receiving SIP messages over UDP for a particular packet group. These UDP paths, however, have no independent status and cannot be separately maintained (i.e., cannot be manually removed or restored).

□

Hardware View

Overview

This feature introduces two protocol handlers, the PHE2 and PH33. However, the terms SIP PH and GQPH are used to indicate that the PHE2 operates as a SIP PH and the PH33 operates as a GQPH, to be consistent with the document unless otherwise noted.

Basic Description

SIP PH Channel Group

The SIP PH channel group (logical protocol handler type) uses the PHE2 protocol handler. The PHE2 is a member of the PH30 family of protocol handlers. The PHE2 supports one 100BaseT paddle board. Table 2-2, "SIP PH Characteristics" (2-24) lists some key characteristics.

Table 2-2 SIP PH Characteristics

Channel Group Type	SIPT
CLI	0x20000
Software Image	PHE2S
Hardware Type	PHE2
Pack Code	TN13
Paddle Board Code	LLE2 (Ethernet)

The LLE2 paddle board is located on the back side of the PSU2 and is mounted directly behind its associated SIP PH. The LLE2 provides the termination for one Ethernet link.

Refer to *Hardware Description*, 235-100-200 for additional information on the PHE2 and LLE2 paddle board.

GQPH Channel Group

The GQPH channel group (logical protocol handler type) uses the PH33 protocol handler. Like the PHE2, the PH33 is a member of the PH30 family of protocol handlers. Table 2-3, "GQPH Characteristics" (2-24) lists some key characteristics.

Table 2-3 GQPH Characteristics

Channel Group Type	GQPH
CLI	0x880

Table 2-3 GQPH Characteristics (continued)

Software Image	PH33Q
Hardware Type	PH33
Pack Code	TN113

Note: The GQPHs are only on the non-DRM/VCDX.

Refer to *Hardware Description*, 235-100-200 for additional information on the PH33.

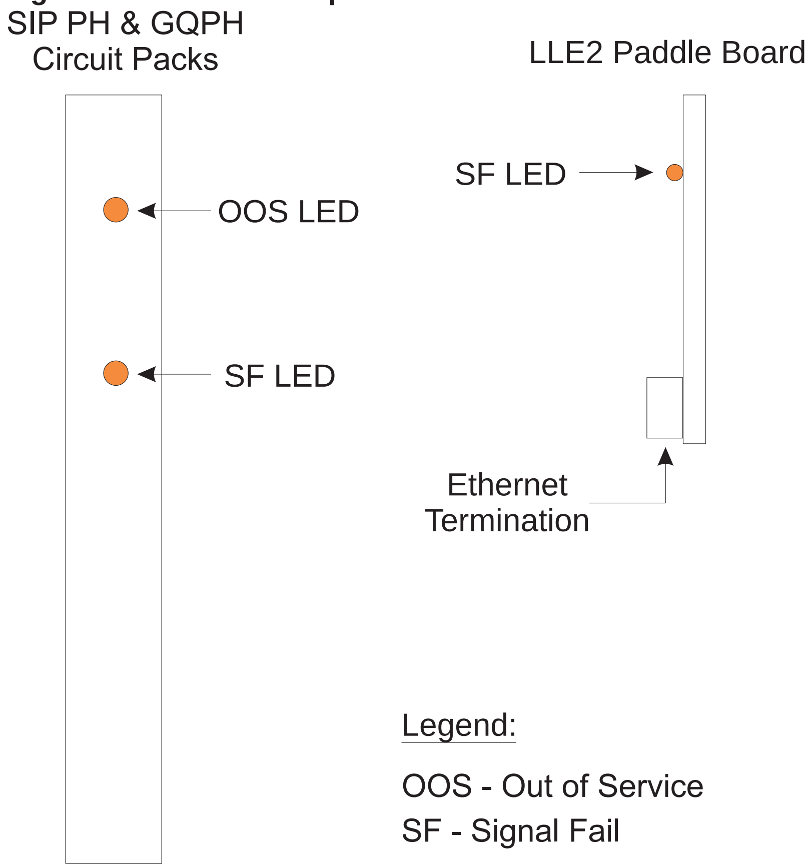
Pack physical characteristics

Each SIP PH or GQPH circuit pack is located in a PH slot as required. They each contain two amber LEDs in the faceplate. Refer to Figure 2-14, "Pack Faceplates" (2-26) for an example of the SIP PH and GQPH faceplates and LEDs. Both the out-of-service (OOS) and signal fail (SF) LEDs are illuminated when the packs are removed from service. The SF LED, on the ***SIP PH only***, is also illuminated when there is no signal from the Ethernet termination.

The LLE2 paddle board is located on the back side of the PSU2. One paddle board is attached to the upper pin field directly behind the corresponding SIP PH. The paddle board is equipped with an SF LED. The SF LED is illuminated when there is no signal from the

Ethernet cable. At the bottom of the paddle board there is a connector for the Ethernet cable. Refer to Figure 2-14, "Pack Faceplates" (2-26).

Figure 2-14 Pack Faceplates

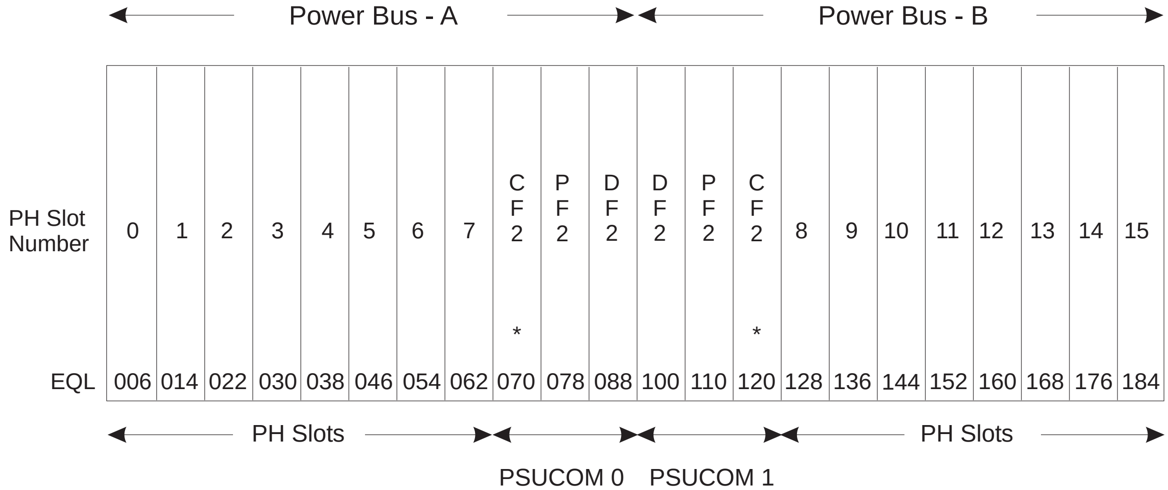


PSU2 shelf arrangement

Figure 2-15, "PSU2 Shelf Arrangement" (2-27) provides the general layout of a PSU2 shelf. The shelf shown is the common shelf (shelf 0). A fully equipped PSU2 consists of five shelves; the common shelf and four growth shelves.

The PSU2 shelves can be located in the following vertical EQLs: 19, 28, 45, 53 and/or 62.

Figure 2-15 PSU2 Shelf Arrangement



* Note: the CF2 packs are located in the common shelf only.

When the shelf is equipped with PHs to support SIP for Packet Trunking, any of the PH slots can be used. If a processor group consists of two SIP PHs, the SIP PHs can either be on the same PSU2 shelf or different shelves. Each shelf that has one or more GQPH channel groups assigned to it should have at least one spare PH33.

PSU2 shelf power

A separate power bus provides power to each half of a PSU2 shelf. Refer to Figure 2-15, "PSU2 Shelf Arrangement" (2-27)

Each shelf receives power through six 10 amp fuses. Specifics on the EQLs and slots powered by each fuse can be found in Table 2-4, "PSU2 shelf power terminations" (2-27). The table also provides a reference to the -48V termination.

Table 2-4 PSU2 shelf power terminations

Power Bus	Shelf EQLs Powered	Slots	Fuse Value	Back Plane Termination
A	006-030	PH 0-3	10 amp	02-028-006
A	038-062	PH 4-7	10 amp	02-060-006
A	070-088	PSUCOM 0	10 amp	02-086-006

Table 2-4 PSU2 shelf power terminations (continued)

Power Bus	Shelf EQLs Powered	Slots	Fuse Value	Back Plane Termination
B	100-120	PSUCOM 1	10 amp	02-108-006
B	128-152	PH 8-11	10 amp	02-134-006
B	160-184	PH 12-15	10 amp	02-166-006

The PHs should be spread across power buses and fuses. Refer to Chapter 4, "Engineering Considerations" for more specific recommendations.

Cabling

Ethernet cables provide the connection between the LLE2 paddle boards (located on the back plane of the unit) and the layer 2 switch or router. The location of each LLE2 paddle board is dependent on the equipage of the SIP PHs. The physical connection is by a Category 5 cable with RJ-45 connectors. The maximum length of an Ethernet 100BaseT cable is 328 feet.

Connecting circuits

This section provides a brief description of the interfaces that connect to the SIP PH and GQPH. Refer to Figure 2-16, "Connecting circuits" (2-29).

SIP PH and GQPH interfaces consist of the following which are part of the PSU2 back plane:

- packet bus (PB),
- control bus (CB), and
- protocol handler data bus (PHDB).

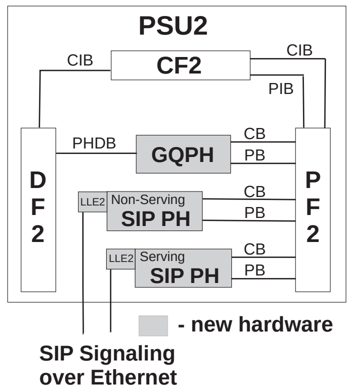
The packet bus provides the packet data interface between the PHs and the packet fanout 2 (PF2). The PHs transmit and receive data packets over the packet bus.

The function of the control bus is to fanout control signals, which are controlled by the CF2, to the PHs and provide PH error status.

The protocol handler data bus provides a data interface between the PHs and the DF2. The PHDB carries time slots between the PH and DF2. The SIP PH does not use the PHDB.

The SIP PH supports an external interface, the LLE2 paddle board. The LLE2 paddle board terminates an Ethernet connection which provides the connection to the layer 2 switch or router.

Figure 2-16 Connecting circuits



Timing

No changes.

Configuration

The SIP PHs are configured as processor groups each having one or two PHs. When configured with two PHs in a processor group, both PHs are active with one being designated as serving and the other PH as non-serving. Refer to section"Serving/Non-Serving" (2-30) for additional information.

The GQPHs are configured as active loadshared PHs with a standby spare PH.

Service impacts to subscriber

Cause	Effect
SIP PH failure	transient calls timeout and abandon; stable calls remain.

Cause	Effect
GSM full initialization with or without pump	most transient calls supported by SIP PHs dropped; stable calls should be recovered.

Service impacts to switch

SIP PH

SIP PH failure when SMP unavailable due to:

Cause	Effect
GSM selective initialization	• inbound and outbound messages discarded, and • associations closed.

Cause	Effect
GSM full initialization with or without pump	Serving SIP PHs attempt to shutdown all established associations before being initialized.

GQPH

GQPH failure when SMP unavailable due to:

Cause	Effect
GSM selective initialization	• inbound messages discarded, and • outbound messages rerouted to alternate GQPH; if no alternate GQPH, messages are discarded.

Note: The GQPHs are only on the non-DRM/VCDX.

Serving/Non-Serving

SIP PHs are configured as processor groups containing one or two SIP PHs. For a processor group of two active SIP PHs, the GSM SMP selects one SIP PH as serving and the other SIP PH as non-serving. For a processor group of one active SIP PH, the GSM SMP selects the single SIP PH as the serving SIP PH.

The serving SIP PH:

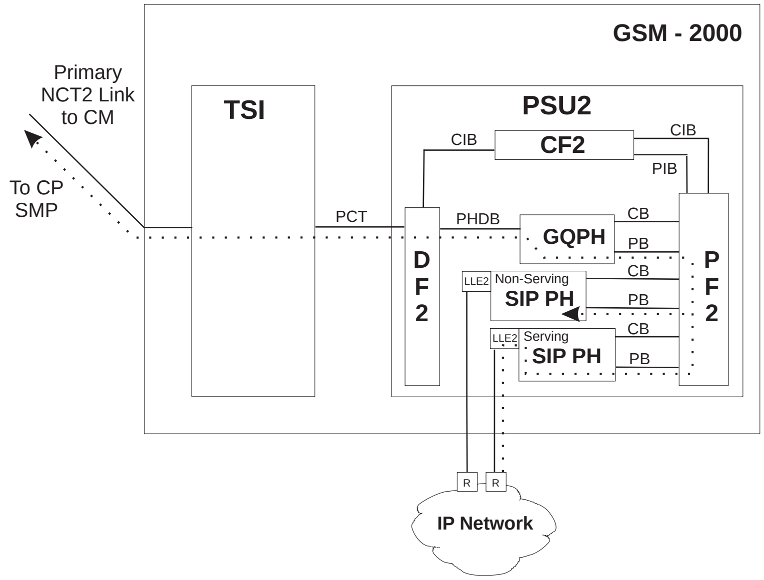
- initiates its local SCTP endpoint and the associations provisioned on that endpoint when SCTP transport is used,
- initiates the SIP interface to UDP paths provisioned on the processor group when UDP transport is used,
- provides the non-serving SIP PH with stable call data to be preserved in case a failure occurs with the serving SIP PH,
- provides the CP SMP with stable call data to address cases that impact both SIP PHs,
- supports address resolution protocol (ARP) functionality, and
- initiates IP pings to confirm IP connectivity.

The non-serving SIP PH:

- Receives stable call data from the serving SIP PH over the PSU2 packet bus interface,
- Does not send or receive packets over the Ethernet interface,
- Is able to detect and report Ethernet link failures.

Figure 2-17, "SIP PH Functionality" (2-31) depicts how the serving SIP PH saves the stable call data.

Figure 2-17 SIP PH Functionality



SIP PH switch over

A SIP PH switch over can be automatic (fail-over) or manual operation (manual switch over). It switches the serving SIP PH to the non-serving SIP PH, and switches the non-serving SIP PH to the serving SIP PH.

Switch over from the serving SIP PH to the non-serving SIP PH occurs in 3 seconds or less.

A 5 minute timer is started following a switch over as a result of auto ping failing in the serving SIP PH. The timer is used to prevent ping failures on the new serving SIP PH from attempting continual switches. However, the timer does not prevent a manual switch over or a switch over due to a failure condition.

When the timer expires, the following can occur.

- If ping is up on the serving SIP PH, nothing occurs.
- If ping is down on the serving SIP PH and the other SIP PH is non-serving, a switch over occurs even if the non-serving SIP PH had a previous known ping failure.
- If ping is down on the serving SIP PH and the other SIP PH is unavailable, no switch over occurs.

The following failure conditions cause the serving SIP PH to do a SIP PH switch over:

- service selection state changes to unavailable (e.g., Ethernet link failure or SIP PH card failure), and
- loss of ping and the timer has expired.

A switch over does not occur when the service selection state of the non-serving SIP PH changes to unavailable.

Service Selection

The service selection state for a SIP PH indicates its ability to support or handle signaling services. The SIP PH may transition to another state automatically or manually by the switch personnel.

There are 3 service selection states:

- ***Serving*** - indicates that a SIP PH is performing the signaling role for the SIP layer, transport layer, and IP layer. If the processor group supports an SCTP near endpoint for the transport layer, then the serving SIP PH establishes the SCTP associations terminated on that endpoint, connecting sockets between SCTP and the SIP layer when the associations are established to the far end. If the processor group supports UDP paths for the transport layer, then the serving SIP PH connects the sockets between UDP and the SIP layer.
- ***Non-Serving*** - indicates that a SIP PH is not performing the signaling role but can take over the signaling role when needed.
- ***Unavailable*** - indicates that a SIP PH cannot perform any signaling activities and cannot take over the signaling role if the serving SIP PH fails.

Table 2-5, “Service Selection State” (2-33) illustrates the valid service selection states allowed based on the status of the PHE2.

Table 2-5 Service Selection State

SERVICE SELECTION STATE	PHE2 STATUS
SERVING	ACTIVE
NON-SERVING	ACTIVE
UNAVAILABLE	OOS
UNAVAILABLE	DEGRADED

The following conditions cause the SIP PH to transition to the unavailable service selection state:

- SM full initialization,
- PH full initialization,
- SIP PH out of service (OOS),
- Ethernet OOS, and
- PSU2 failures.

When a SIP PH transitions from the service selection state of serving or non-serving to the unavailable state due to a non-manual request, a major alarm is reported.

When a SIP PH transitions from the service selection state of serving to the unavailable state and the mate SIP PH is in the unavailable state, a critical alarm is reported.

Table 2-6, “Service Selection State Combinations” (2-34) illustrates the allowed service selection state combinations for a processor group with two SIP PHs.

Table 2-6 Service Selection State Combinations

SIP PH 0	SIP PH 1
SERVING	NON-SERVING
NON-SERVING	SERVING
UNAVAILABLE	UNAVAILABLE
UNAVAILABLE	SERVING
SERVING	UNAVAILABLE

However, for a processor group with only one SIP PH, the allowed service selection states are serving and unavailable.

☐

Security View

Overview

The bearer and signaling interfaces on the IP core network may or may not be trusted and secure depending on the service provider's network implementation. This section provides security information.

Bearer Security

Additional security features were developed on the OIU-IP. The OIU-IP uses the real time transfer protocol (RTP) to carry voice traffic on protected packet over SONET (POS) fiber optic links.

The security features for the OIU-IP bearer interface include:

- point to point protocol (PPP) link scrambling,
- valid user datagram protocol (UDP) port range matching,
- dynamic packet filtering of voice RTP traffic,
- invalid packet discard for protocol errors,
- internet control message protocol (ICMP) message processing controls,
- provisionable thresholds for triggering selected minor alarms when detecting protocol violations, and
- detection of flood attack on links, and routing of new calls away from such links during duration of the attack.

OIU-IP bearer network security is implemented in two main areas, the packet field programmable gate array (PFPGA) on the OFI-IP and the router to which the OFI-IP is connected. The router connected to an OFI-IP performs basic filtering functionality and guards the OC-3c intra-office link. The router should be capable of rough filtering non-relevant traffic away from the OFI-IP interface. A router which permits configurable packet filter policies based on IP destination protocol and is capable of rate limiting to minimize the effect of ping floods on the OC-3c link is recommended.

The PFPGA monitors the bit-stream for a large variety of protocol violations and discards nonconforming or malicious packets. Discarded packets are counted, and the counts are listed in various digital performance monitoring (DPM) and traffic measurement reports. A subset of the DPM counts can be provisioned with a minor alarm at DPM threshold crossing.

The *OIU-IP Interface Specification,* 235-900-316 document explains OIU-IP security in greater detail.

SCTP Security

The *5ESS*® switch uses SCTP layer security according to the SCTP standards. SCTP provides for protection against:

- flooding,
- blind masquerade,
- improper monopolization of services, and
- fraud and repudiation.

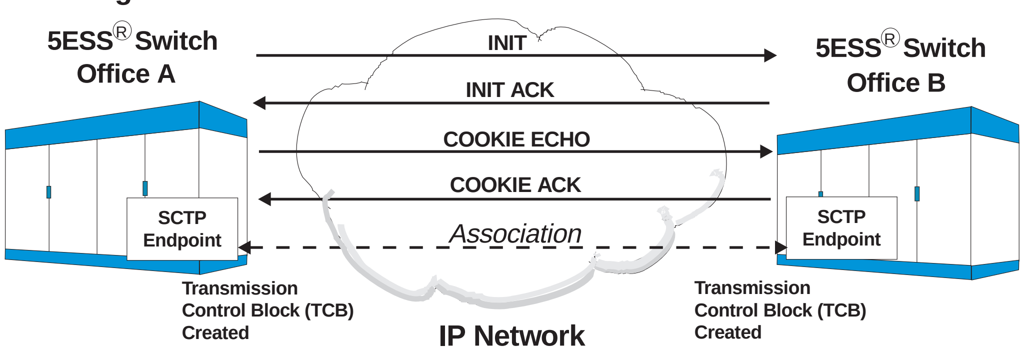
SCTP uses a cookie mechanism which is started during initialization to provide protection against security attacks. This is an advantage over TCP and UDP because they do not have this application function.

Cookie Mechanism

Figure 2-18, "Cookie Mechanism" (2-36) illustrates the cookie mechanism.

1. The client, office A, sends a connection request (INIT) to the server.
2. The server, office B, builds a cookie (INIT ACK) containing TCB information and sends it to the client.
3. The client returns the TCB information to the server (COOKIE ECHO).
4. The server validates the cookie and uses it to rebuild the TCB that it returns to the client (COOKIE ACK).

Figure 2-18 Cookie Mechanism



The advantage of the cookie mechanism is that the server does not reserve memory or resources until a COOKIE ECHO message is received from the client. This protects the server from overload during blind attacks.

A blind attack occurs when a client is sending a client IP address different from its own to the server. The server returns the cookie to the invalid client IP address instead of the attacking client. The invalid client will drop the message instead of returning a COOKIE ECHO message. Since the server never receives a COOKIE ECHO message, memory and resources are not allocated and overload is avoided.

□

3 Call Flow

Overview

Purpose

The purpose of this chapter is to describe the Session Initiation Protocol (SIP) call flow.

The purpose of this chapter is to describe possible Session Initiation Protocol (SIP) call flows.

Call Flow Overview

When the *5ESS*® switch determines a call must be routed to another switch, the call processing programs in the *5ESS*® switch determine a path to route the call. If the switch selects an IP network path, call processing programs use Session Initiation Protocol (SIP) signaling to establish the call. The switch converts incoming in-band or out-of-band signaling messages to SIP messages so Public Switch Telephone Network (PSTN) calls can traverse an IP network. In this situation, the switch is functioning as an Originating Packet Switch (OPS) at the boundary between the PSTN and the IP network.

At the far-end switch, the call is typically completed to a time division multiplexing (TDM) circuits. When a call is completed to a TDM circuit at the terminating *5ESS*® switch, the appropriate PSTN outgoing signaling message is formulated from the received SIP message.

The *5ESS*® switch can also terminate calls that arrive from an IP network, in which case it converts incoming SIP signaling message to in-band or out-of-band signaling in order to complete the call on a circuit path in the PSTN. In this situation, the switch is functioning as a Terminating Packet Switch (TPS) at the boundary between the PSTN and the IP network.

Many call flows are possible, depending on the following factors:

- network architecture,
- function of the *5ESS*® switch in the network,
- SIP protocol options supported by other SIP-enabled switches which the *5ESS*® switch connects via a SIP packet group, and the function of those far-end switches in the PSTN and/or IP network,
- transport layer used to carry the SIP messages in the IP network,
- provisioned mapping rules for signaling message conversion between PSTN and SIP protocols for the packet group on the *5ESS*® switch, and
- SIP protocol options provisioned for the packet group on the *5ESS*® switch.

□

Network Architecture

The *5ESS*® switch can function as either an OPS or TPS in different network architectures:

- **Packet Trunking:** The *5ESS*® switch in the PSTN, connected to another SIP-enabled switch in the PSTN by means of SIP packet trunking (call originates and terminates in PSTN, but traverses the IP network between the PSTN endpoints). This network architecture is illustrated in Figure 3-1, "Packet Trunking" (3-3).
- **PSTN Gateway:** The *5ESS*® switch in the PSTN connected to an IP Telephone Application Server (TAS) TPS in the IP network, with the *5ESS*® switch serving as a PSTN Gateway for calls originating in the PSTN and terminating in the IP network, or originating in the IP network and terminating in the PSTN. This network architecture is illustrated in Figure 3-2, "PSTN Gateway" (3-4).

Figure 3-1 Packet Trunking

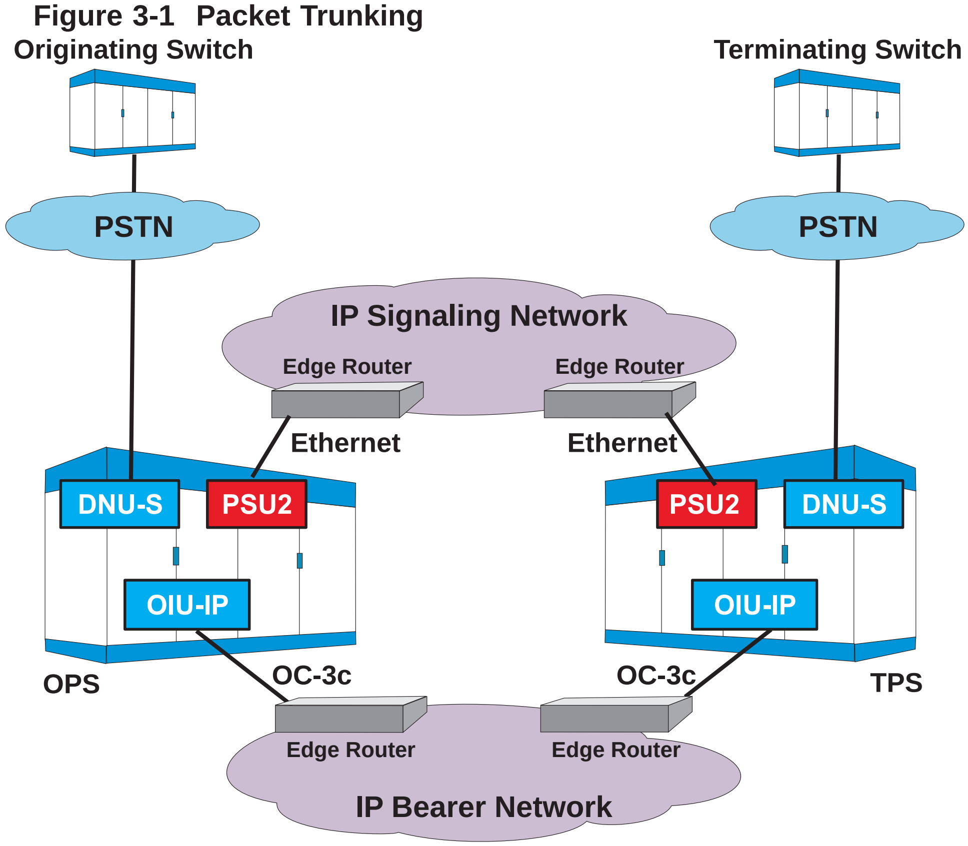
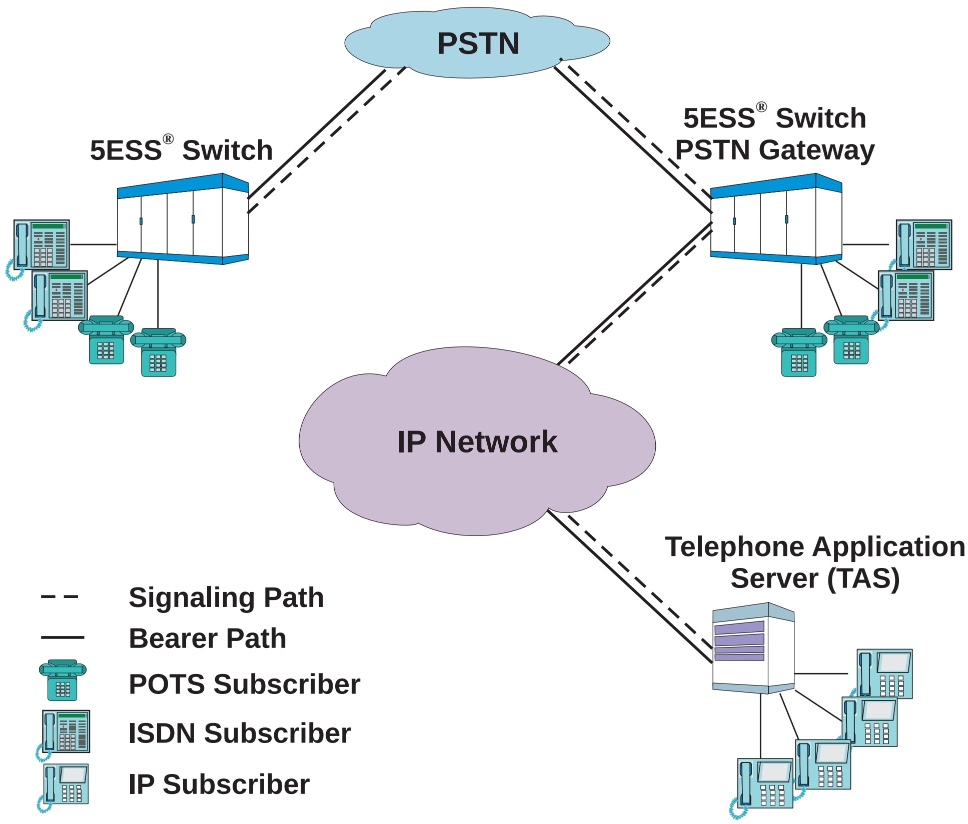


Figure 3-2 PSTN Gateway



Message conversion between PSTN and IP protocols

PSTN signaling messages are converted to SIP messages using translation and encapsulation. For outgoing messages, the SIP protocol handler (PH) in the *5ESS*® switch constructs a SIP message in a standard, external format using information provided by the SIP call processing SM.

The SIP message contains the following:

- information used to route the message in the IP network,
- information describing the message type and content, and
- data associated with the particular application.

For incoming messages, the SIP PH extracts information from the SIP message and delivers the data to the SIP call processing SM. The SIP call processing SM passes the information to the SM that generates PSTN signaling messages and completes the call using TDM circuits.

The details of how the call signaling messages are converted to and from SIP are controlled by a number of provisionable parameters on the *5ESS*® switch. Provisionable parameters control what signaling information from *5ESS*® switch call processing is (or is not) translated into SIP headers; whether (or not) encapsulated ISUP is included in SIP headers at the OPS; and what information from the SIP signaling messages is (or is not) extracted from the SIP headers and/or encapsulated ISUP to be delivered to call processing at the TPS. The setting of these parameters for each packet group on the *5ESS*® switch is dependent on the SIP functionality supported by the switch at the far end of the packet group.

Notes:

1. Refer to the *Session Initiation Protocol (SIP) - Interface Specification*, 235-900-344, document for more detailed information on SIP messages and interworking to and from TDM messages.
2. Refer to System View in Chapter 2 of this document for more information about the *5ESS*® switch elements involved in the intra-switch messaging between the SIP PH and the Call Processing SM.
3. Refer to Chapter 5 of this document for details of provisioning procedures for the Recent Change views that affect call flows, particularly the provisioning of SIP Packet Groups (RC/V 5.71), SIP Parameter Sets (RC/V 5.82), and mapping rules for interworking between SIP and PSTN signaling (RC/V 5.83).

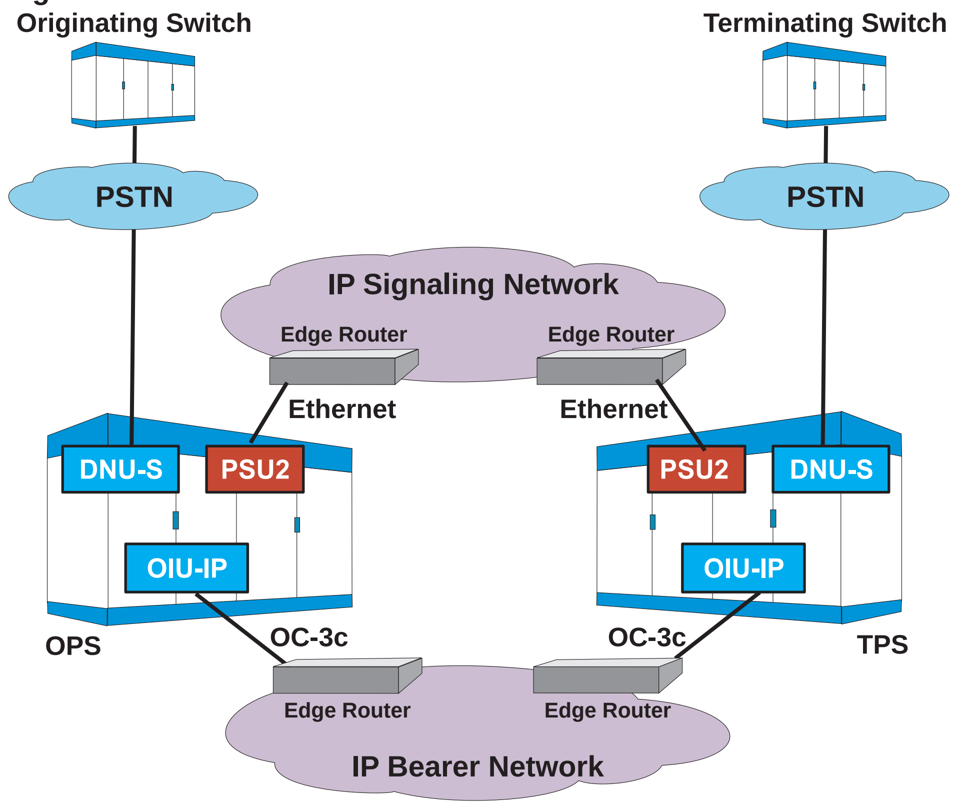
Elements

The initial application for SIP signaling is in tandem/toll offices. Considering this, there are five key network elements that a call traverses. They are:

- Originating Switch
- Originating Packet Switch (OPS)
- Routers
- Terminating Packet Switch (TPS)
- Terminating Switch

[Figure 3-3, “Network Elements” (3-6)](#), illustrates how the different network elements interface to each other.

Figure 3-3 Network Elements



Originating Switch

The originating switch terminates the calling party and provides an interface to the PSTN.

Originating Packet Switch (OPS)

The OPS serves as a gateway between the PSTN and an IP network. At the OPS, the incoming circuit call is routed out of the office to an IP bearer resource using SIP. Any circuit peripheral can be used for the incoming call.

Three SMP processes are involved in a SIP call:

- the SMP process for the circuit call (ISUP terminal process)
- the SMP process for the SIP call processing SM (SIP terminal process), and
- the SMP process with the OFI-IP for the bearer connection (bearer terminal process).

All three processes could be located on the same SM-2000 or they could be located in three different SM-2000s. Messaging between the SMPs that host the processes is done over the quad link packet switch (QLPS) network.

Routers

Control the routing of packets between the OPS and TPS and provides alternate paths when necessary. A router is able to the read network layer, IP, addresses of the transmitted packets and only forwards those addressed to another network. Routers do not examine application layer messages like SIP.

Terminating Packet Switch (TPS)

The TPS serves as a gateway between the PSTN and an IP network. At the TPS, the incoming SIP call is routed out of the office to the TDM network using TDM signaling. Any circuit peripheral can be used for the outgoing call.

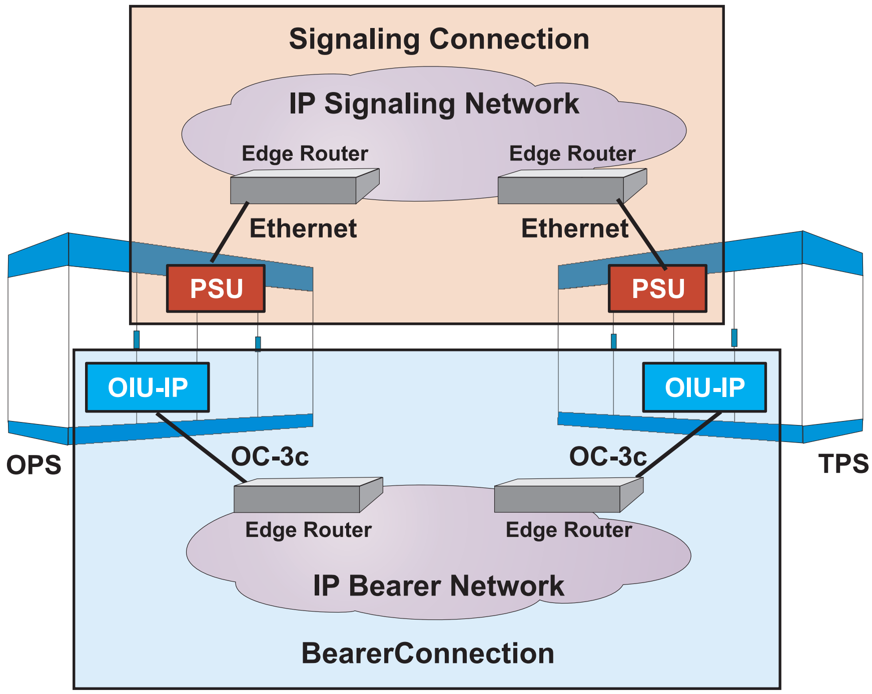
Terminating Switch

The terminating switch terminates the called party and interfaces to the PSTN.

Connections

There are two network connections, bearer and signaling. The two connections can be made through separate networks or through the same network. Figure 3-4, "Network Connections" (3-8), illustrates connections to separate IP networks.

Figure 3-4 Network Connections



Signaling connection

The signaling connection is only provided by an SM-2000. A 100BaseT Ethernet connection passes the signaling messages between the edge router and the SIP PH in the packet switching unit model 2 (PSU2).

Bearer connection

The bearer connection is only provided by an SM-2000 that has optical facility interface-internet protocol (OFI-IP) boards in the OIU. It is the OFI-IP in an OIU that provides the synchronous to asynchronous conversion between TDM bearer flows and IP bearer flows.

□

High-Level Call Flow

SIP OPS Call Setup

This section describes the steps for setting up a SIP call on a *5ESS*® switch OPS from a high-level view. It presumes that the incoming circuit call to be terminated over a SIP packet group is an ISUP trunk call, although other circuit originations can terminate to SIP as well.

1. An ISUP IAM message from the far switch in the PSTN arrives at the *5ESS*® switch OPS. Call processing determines that the incoming ISUP call should be routed out over the IP network to another switch.
2. The OPS selects and allocates an IP address and port on an OFI-IP to be the bearer for the call.
3. The OPS formulates a SIP INVITE message. The SIP INVITE message requests that the TPS allocate resources for the call. This message includes:
   - the IP address of OPS signaling point,
   - the IP address of TPS signaling point,
   - a description of the requested bearer session, including the IP address and port number for the OPS bearer resource,
   - information about the SIP methods and procedures supported by SIP at the OPS, and
   - call information from the ISUP IAM, such as the called and calling party information.
     Note: The call information from the IAM is translated into the SIP headers according to the PSTN-to-SIP mapping rules provisioned for the selected SIP packet group, and the ISUP IAM may be encapsulated in the SIP message or not, according to the provisioning of the ISUP encapsulation parameter for the packet group.
4. The OPS sends the SIP INVITE to the TPS, and waits for a response. If the transport layer is UDP, the SIP layer at the OPS retransmits the INVITE until a response is received. If the transport layer is SCTP, which is a reliable, connection-oriented protocol, the SIP layer does not retransmit the INVITE.

5. If the transport layer is UDP, the first provisional response is likely to be a ″100 TRYING″ message unless the TPS is a *5ESS*® switch and the transport layer is SCTP. The ″100 TRYING″ message indicates that the far end has received the INVITE, so that retransmissions can be halted, but provides no other information for call setup. The OPS continues to wait for a response containing the necessary session description and other signaling information.
6. Depending on the nature of the call, the next response received (or the first response, when the transport layer is SCTP), is likely to be a provisional response such as ″180 RINGING″ or ″183 SESSION PROGRESS″, which contains the bearer session description of the far end, as well as other call signaling information, possibly including encapsulated ISUP, if encapsulated ISUP was included in the original INVITE.
7. There may, or may not, be additional SIP messages exchanged between the OPS and TPS before the call is answered, depending on the nature of the call and the provisioned options for SIP methods and procedures allowed and/or supported at each end, which are communicated and agreed upon between the OPS and TPS, by means of parameters in SIP headers in the INVITE/18X SIP message exchange. The options that might lead to additional messages being exchanged before the call is answered include support for SIP preconditions procedures (allowed only when the transport layer is SCTP), and support for reliable provisional responses (PRACK).
8. When the call has been answered, the OPS receives a 200 OK INVITE final response.
9. The OPS acknowledges the 200 OK INVITE with a SIP ACK message. Whether the transport layer is UDP or SCTP, the SIP protocol requires the 200 OK INVITE final response to be retransmitted by the TPS until the ACK is received, so the OPS must respond to any retransmissions of the 200 OK INVITE with retransmissions of the ACK message.
10. After the 200 OK INVITE is received and the ACK is sent, the call proceeds to the ″talking″ state, and call setup is complete.

At this point the bearer path for the call is available to carry call data (e.g., PCM voice) between the calling and called party. The path remains available until either party terminates the call.

SIP TPS Call Setup

The following is a high-level view of call setup from the viewpoint of a *5ESS*® switch TPS:

1. The *5ESS*® switch TPS receives a SIP INVITE from a far switch.
2. If the transport layer is UDP, the TPS immediately responds with a ″100 TRYING″ message to inform the OPS that the INVITE has been received, so that INVITE retransmissions may cease. If retransmitted INVITEs are received, the TPS responds by retransmitting the latest INVITE response sent.
3. The TPS extracts the call signaling data from the SIP headers and/or the encapsulated ISUP IAM (if any) in the SIP INVITE, according to the provisioned SIP-to-PSTN mapping rules.
4. The TPS selects and allocates an IP address and port on an OFI-IP to be the bearer for the call.
5. The TPS uses the call signaling data to route the call to an outgoing ISUP trunk. The call signaling data extracted from the SIP message is included in an ISUP IAM to be sent the PSTN switch at the far end of the ISUP trunk.
6. The TPS formats and sends a provisional response to the OPS. The exact response, such as ″183 SESSION PROGRESS″ or ″180 RINGING″, depends on the nature of the call, the methods and procedures supported by the OPS as indicated in the incoming INVITE, and the methods and procedures supported by the TPS, according to the SIP parameters provisioned for the packet group over which the INVITE was received. The response includes the bearer session information determined by the TPS, and may or may not include an encapsulated ISUP response from the switch at the far end of the ISUP trunk group in the PSTN, depending on whether the received INVITE included encapsulated ISUP.
7. There may, or may not, be additional SIP messages exchanged between the OPS and TPS before the call is answered, depending on the nature of the call and the provisioned options for SIP methods and procedures allowed and/or supported at each end, which are communicated and agreed upon between the OPS and TPS by parameters in SIP headers in the INVITE/18X SIP

message exchange. The options that might lead to additional messages being exchanged before the call is answered include support for SIP preconditions procedures (allowed only when the transport layer is SCTP), and support for reliable provisional responses (PRACK).

8. If PRACK procedures are supported by both the OPS and the TPS, the TPS will retransmit the 18X provisional response until a PRACK is received to acknowledge it, or until the call is answered. This retransmission of provisional responses is performed by the SIP layer independent of transport layer, over either SCTP or UDP. When PRACK procedures are enabled, the TPS will not send any additional provisional responses, and will not accept any new requests, until the PRACK is received to acknowledge the last provisional response. If PRACK procedures are not supported by both ends, the TPS will not retransmit the 18X response when the transport layer is SCTP, and will only retransmit the 18X response as a result of receiving a retransmitted INVITE request from the OPS when the transport layer is UDP.
9. When an ISUP message is received indicating that the call has been answered at the other end of the ISUP trunk in the PSTN, the TPS formats and sends a 200 OK INVITE final response. The TPS retransmits the final response, independent of transport layer, until it receives an ACK from the OPS.
10. After the 200 OK INVITE is sent and the ACK is received, the call proceeds to the ″talking″ state, and call setup is complete.

SIP Call Tear Down

This section describes tearing down of a SIP call, from a high-level view.

Originating Packet Switch:

From the viewpoint of a *5ESS*® switch OPS, when the calling party in the PSTN goes on-hook:

1. The OPS receives an ISUP REL message from the switch at the other end of the ISUP trunk in the PSTN.
2. The OPS formulates a SIP BYE messages and forwards it to the TPS. The OPS also tears down the call and releases resources dedicated to the call.

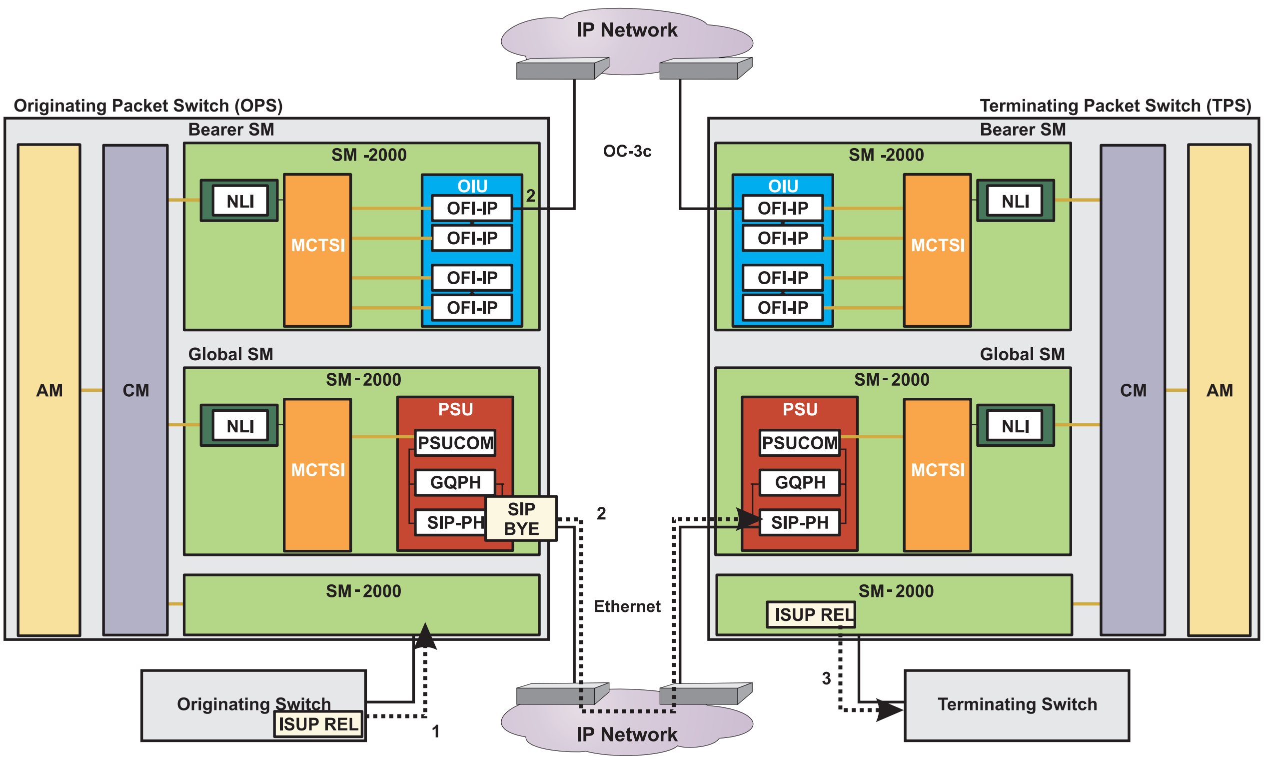
Terminating Packet Switch:

From the viewpoint of the *5ESS*® switch TPS, when the calling party (either in the IP network if the TPS is a PSTN Gateway, or in the PSTN for SIP packet trunking between PSTN offices) goes on-hook:

1. The TPS receives a SIP BYE message.
2. The TPS formulates an ISUP REL message and forwards it to the terminating switch. The TPS also tears down the call and releases resources dedicated to the call.

Tearing Down Call Path

This section overviews tearing down of a SIP call from a high-level view. In this scenario, the calling party ended the call by going on-hook.



1. The calling party goes on-hook. The originating switch send an ISUP REL message to the OPS to notify the OPS of the state change.
2. The OPS formulates a SIP BYE messages and forwards it to the TPS. The OPS also tears down the call and releases resources dedicated to the call.

3. The TPS receives the SIP BYE message. The TPS formulates an ISUP REL message and forwards it to the terminating switch. The TPS also tears down the call and releases resources dedicated to the call.
4. The call flow is now complete.

Message Flows

Figure 3-5, "Successful Call Completion Message Flow" (3-15) illustrates ISUP and SIP messages sent between network elements in order to successfully setup and tear down a call between two subscribers.

Figure 3-5 Successful Call Completion Message Flow

```
Originating                              Terminating
       Switch     OPS                   TPS      Switch
         | --IAM-> |                     |          |
         |         | ------INVITE-----> |          |
         |         |                     | --IAM-> |
         |         |     183 SESSION     |          |
         |         | <----PROGRESS----- |          |
         |         | ------UPDATE-----> |          |
         |         | <-200 OK (UPDATE)- |          |
         |         |                     | <-ACM-- |
         |         | <---180 RINGING--- |          |
         | <-ACM-- |                     |          |
         |         |                     | <-ANM-- |
         |         | <-200 OK (INVITE)- |          |
         | <-ANM-- |                     |          |
         |                                          |
         |          Talking Path Available          |
         |                                          |
         | --REL-> |                     |          |
         |         | --------BYE------> |          |
         |         |                     | --REL-> |
         |         | <--200 OK (BYE)--- |          |
```

Overview

There are many possible message flows for successful SIP calls, for various network configurations and SIP protocol options. This section illustrates three likely message flows, with different configurations and options provisioned.

Message Flow 1

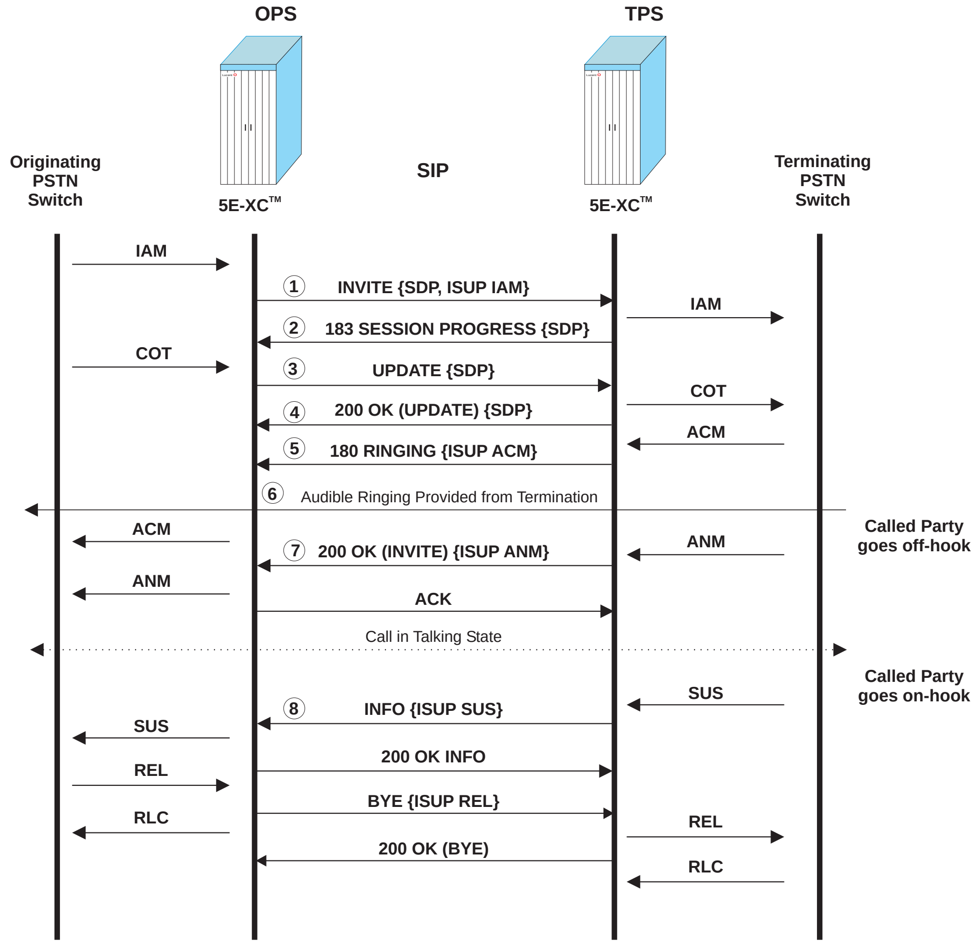
This message flow is a straightforward packet trunking scenario between a *5ESS*® switch OPS and *5ESS*® switch TPS in the PSTN, where the call originates from an ISUP trunk in the PSTN at the OPS, and terminates to an ISUP trunk in the PSTN at the TPS. Refer to Figure 3-6, " *5ESS*® Switch OPS - *5ESS*® Switch TPS Message Flow" (3-17).

The provisioning at both OPS and TPS which affects the message flow and message content is as follows:

- Transport Layer: SCTP (RC/V 5.71 ASSOC SET NAME provisioned),
- Preconditions: Yes (RC/V 5.71 SIP-T PRECOND=Y),
- Local Audible: No (RC/V 5.71 LOCAL AUDIBLE=N),
- Encapsulated ISUP: Yes (RC/V 5.82 ISUP ENCAPSULATION=Y),
- PRACK procedures: No (RC/V 5.82 PRACK ENABLED=N),
- PSTN-to-SIP mapping rules at OPS: [***don't*** map optional SIP headers] RC/V 5.83:
  - REQUEST URI ADDR=BASIC
  - REQUEST URI CIC=BASIC
  - REQUEST URI CSEL=NOTMAPPED
  - TO HEADER=BASIC
  - FROM ADDR=BASIC
  - FROM NAME=NOTMAPPED
  - FROM OLI=NOTMAPPED
  - P-ASSERTED ADDR=NOTMAPPED
  - P-ASSERTED NAME=NOTMAPPED
  - DIVERSION=NOTMAPPED
  - MAX FORWARDS=BASIC
- SIP-to-PSTN mapping rules at TPS: [***use*** encapsulated ISUP IAM] RC/V 5.83:
  - CALLING PARTY INFO=ISUPMIME
  - CALLED PARTY NUMBER=ISUPMIME
  - TRANSIT NETWORK SELECTION=ISUPMIME
  - REDIRECTING INFO=ISUPMIME
  - ORIGINATING LINE INFO=ISUPMIME

- HOP COUNTER=ISUPMIME
- CARRIER SELECTION=ISUPMIME

Figure 3-6 *5ESS*® Switch OPS - *5ESS*® Switch TPS Message Flow



Note: All SIP messages have the common SIP headers used to identify and route SIP messages and associate them with particular transactions within calls, e.g.: either Request URI or response status line, To, From, Call-ID, CSeq, Via, Contact, Length, Max-Forwards, etc. The following message flow description does not explicitly list these. For more detailed examples, refer to the *Session Initiation Protocol (SIP) - Interface Specification*, 235-900-344, document.

This message flow starts with an ISUP Initial Address Message (IAM) received at the OPS, which routes the call to a SIP packet trunk and sends a SIP INVITE message to the TPS. The ISUP

IAM for this scenario has the COT indicator set, indicating that an ISUP COT message will be received for the ISUP trunk.

1. The SIP INVITE includes:
   - SDP offer with bearer session description for the OPS,
   - encapsulated ISUP Initial Address Message,
   - SIP Allow header indicating support for these methods: INVITE, ACK, BYE, CANCEL, UPDATE, INFO, OPTIONS,
   - SIP Require header indicating that Precondition procedures are required.

   Due to the provisioning of the PSTN TO SIP MAPPING RULES on RC/V 5.83, some call information is not populated in SIP headers, even if the corresponding ISUP parameter is present in the ISUP IAM:
   - no carrier selection information in Request URI,
   - no calling name information in the From header,
   - no originating line information in the From header,
   - no P-Asserted-Identity header, and
   - no Diversion headers.
2. When the TPS has allocated bearer resources, it sends a 183 SESSION PROGRESS which includes:
   - SDP answer with bearer session description for TPS,
   - SIP Allow header indicating support for these methods: INVITE, ACK, BYE, CANCEL, UPDATE, INFO, OPTIONS,
   - SIP Require header indicating that Precondition procedures are required.
3. The OPS sends a SIP UPDATE message as part of the preconditions procedures, after COT has been received from the PSTN and 183 SESSION PROGRESS with SDP has been received from the TPS. The UPDATE includes an SDP offer with the same bearer IP address and port information as the SDP in the original INVITE, but with different quality-of-service attributes.
4. 200 OK UPDATE includes an SDP answer with the same bearer IP address and port as the SDP in the 183 SESSION PROGRESS, but with different quality-of-service attributes.

5. When an ISUP Address Complete message is received from the terminating switch in the PSTN, the TPS sends a SIP 180 RINGING provisional response, which includes:
   - encapsulated ISUP Address Complete Message, and
   - the same SIP Allow and Require headers as the 183 SESSION PROGRESS.
6. Audible ringing is provided by the terminating switch in the PSTN.
7. When the TPS receives an ISUP Answer message, it sends a SIP 200 OK INVITE response, which includes the encapsulated ISUP ANM.
8. When the called party goes on-hook first, the TPS receives an ISUP Suspend message, which it sends in a SIP INFO message with the ISUP SUS message.
   Note: When the calling party goes on hook, the OPS receives an ISUP Release and sends a SIP BYE message with the encapsulated ISUP REL Message. The call resources are then released.

Message Flow 2

In this message flow, the *5ESS*® switch is acting as a PSTN Gateway, handling a call that originates from subscriber on a TAS in the IP network and terminates to a subscriber on another switch in the PSTN. The *5ESS*® switch is the TPS in this scenario, and the TAS is the OPS. It is presumed in this scenario that the TAS has a directly provisioned route to the *5ESS*® switch PSTN Gateway, so there are no proxies or other intermediate nodes between the *5ESS*® switch PSTN Gateway and the TAS. Refer to Figure 3-7, " *5ESS*® Switch PSTN Gateway TPS and TAS OPS Message Flow" (3-21).

The provisioning at the *5ESS*® switch PSTN Gateway which affects the message flow and message content is as follows:

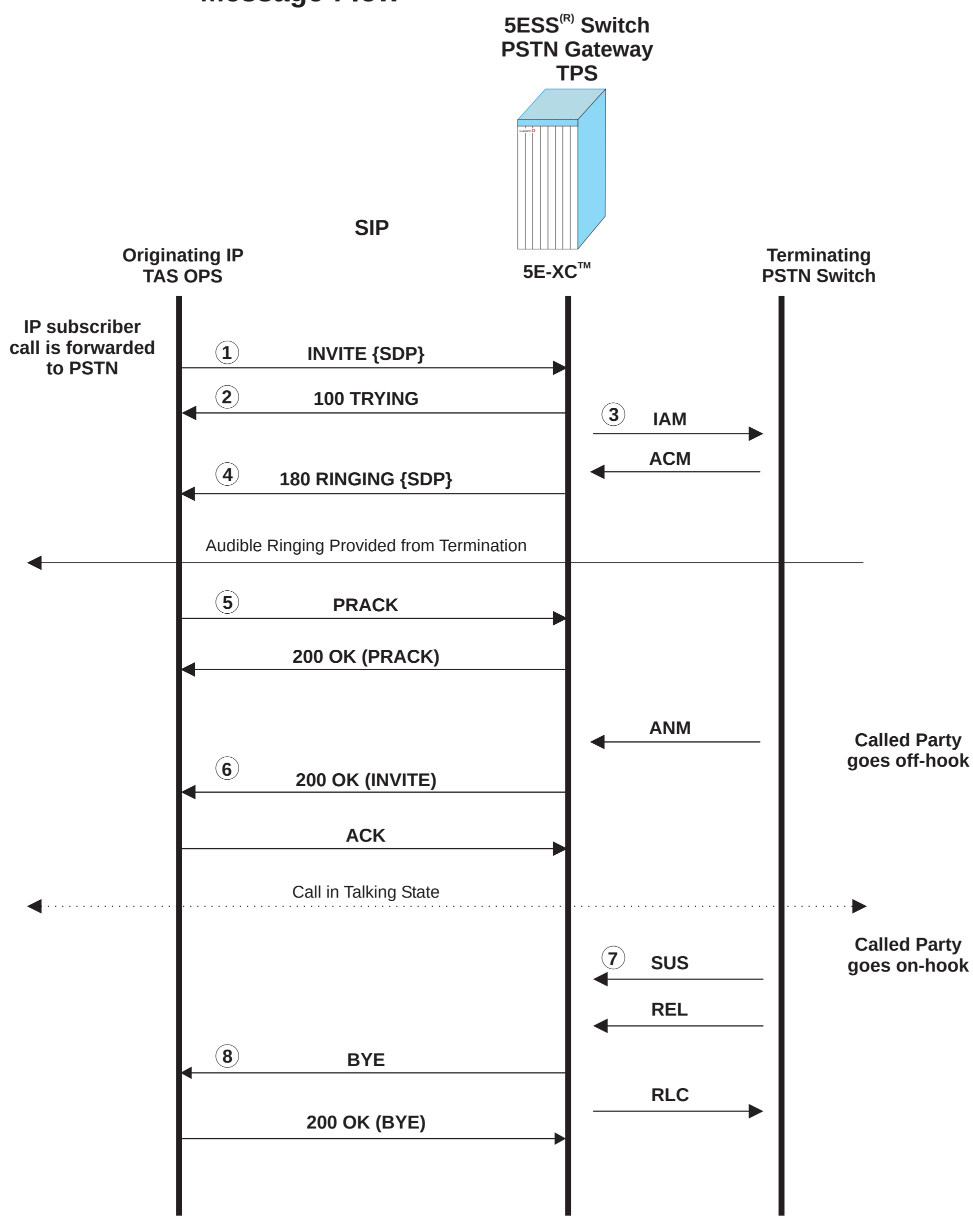
- Transport Layer: UDP (RC/V 5.71 UDP PATH provisioned),
- Preconditions: No (RC/V 5.71 SIP-T PRECOND=N),
- Encapsulated ISUP: No (RC/V 5.82 ISUP ENCAPSULATION=N),
- PRACK procedures: Yes (RC/V 5.82 PRACK ENABLED=Y), and

- SIP-to-PSTN mapping rules at TPS: [extract call information from SIP headers]
  RC/V 5.71: SIP PSTN SET NAME=DEFAULTNOISUP,
  RC/V 5.83:
  - CALLING PARTY INFO=PIDENT,
  - CALLED PARTY NUMBER=BASIC,

- TRANSIT NETWORK SELECTION=BASIC,
- REDIRECTING INFO=DIVERSION,
- ORIGINATING LINE INFO=FROM,
- HOP COUNTER=MAXFORWARD, and
- CARRIER SELECTION=BASIC.

- PSTN to SIP mapping rules do not apply to this example, they only apply to OPS scenarios.

Figure 3-7 *5ESS*® Switch PSTN Gateway TPS and TAS OPS Message Flow



Note: All SIP messages have the common mandatory SIP headers used to identify and route SIP messages and associate them with particular transactions within calls, e.g.: either Request URI or response status line, To, From, Call-ID, CSeq, Via, Length, Max-Forwards, etc. The following message flow description does not explicitly list these. For more detailed examples, refer to the *Session Initiation Protocol (SIP) - Interface Specification*, 235-900-344, document.

The following steps provide a description of the messages between the Originating IP TAS, *5ESS*® switch PSTN Gateway and Terminating PSTN switch.

1. The SIP INVITE includes:
   - SDP offer with bearer session description for originating IP subscriber,
   - SIP Allow header indicating support for at least these methods: INVITE, ACK, BYE, CANCEL, UPDATE, INFO, OPTIONS, PRACK,
   - SIP Supported header indicating support for reliable provisional responses (indicated by ″100rel″ parameter), and
   - SIP P-Asserted-Identity header, with calling party information.

   If the call was forwarded one or more times within the IP network, before ultimately being forwarded to a PSTN subscriber, the SIP INVITE from the TAS OPS may also include a SIP Diversion header for each instance of call forwarding.
   If there are presentation restrictions on the calling party information, the INVITE may include a Privacy header.
   The TAS OPS may include Originating Line Information (OLI),

appended as an `isup-oli=` parameter in the From header. The TAS OPS may include a Carrier ID Code and Carrier Selection Information for the origination, appended as `cic=` and `csel=` parameters on the username portion of the SIP Request URI and the SIP To header.

2. Because the transport layer is UDP, the SIP layer at the TAS OPS will retransmit the INVITE until it receives a response, so the *5ESS*® switch TPS will send a 100 TRYING response immediately after receiving the INVITE to indicate to the OPS that it can cease retransmissions. The *5ESS*® switch TPS will retransmit the 100 TRYING in response to INVITE retransmissions it receives.

3. When the *5ESS*® switch PSTN Gateway has selected an ISUP trunk over which the call is to be routed to the terminating PSTN switch, it constructs an ISUP IAM message for the PSTN side of the call. Since there was no embedded ISUP IAM in the INVITE from the IP TAS, the PSTN Gateway TPS must map information from the SIP headers in the INVITE into ISUP parameters in the IAM, according to the provisioning rules for SIP-to-PSTN mapping on RC/V 5.83. Some of the mappings, given the provisioning described for this scenario, are:
   - SIP Request URI to ISUP Called Party information,
   - SIP P-Asserted-Identity and Privacy headers to ISUP Calling Party information and presentation restrictions,
   - SIP Max-Forwards header to ISUP Hop Counter parameter,
   - SIP `cic=` carrier code treated like a received ISUP TNS in sunsequent switch translations,
   - SIP `csel=` parameter to ISUP Carrier Selection Information,
   - SIP `isup-oli=` parameter to ISUP Originating Line Information , and
   - First and last SIP Diversion headers to ISUP Redirecting information parameters.

4. When the TPS has allocated SIP bearer resources and received an ISUP Address Complete Message (subscriber free) from the terminating PSTN switch, it sends a SIP 180 RINGING provisional response, which includes the following:
   - SDP answer with bearer session description for TPS end of IP bearer,
   - SIP Allow header indicating support for these methods: INVITE, ACK, BYE, CANCEL, UPDATE, INFO, OPTIONS, PRACK,
   - SIP Require header indicating that the call will use reliable provisional responses (indicated by ″100rel″ parameter), and
   - SIP Rseq header with a sequence number specific to this provisional response. (The CSeq number for the INVITE transaction is the same for all provisional and final responses, so an additional sequence number is required to uniquely identify one provisional response for reliable transport.)

   Because the OPS and TPS have agreed to require reliable provisional responses, the TPS will retransmit the 180 RINGING until it receives a PRACK message acknowledging receipt by the OPS.
5. When the TPS receives a PRACK request containing a SIP RAck header with the RSeq of the 180 RINGING, it stops retransmitting the 180 RINGING message and sends a 200 OK PRACK to acknowledge receipt of the PRACK. Because the transport layer is UDP, the OPS will retransmit the PRACK until it receives the 200 OK PRACK. If the TPS receives retransmissions of the PRACK, it will respond with a retransmission of the 200 OK PRACK.

   Note: The TPS will not send any more provisional responses, and will not accept any more requests from the OPS, until the 180/PRACK/200 OK PRACK exchange has been completed.
6. When the PSTN Gateway TPS receives an ISUP ANM from the terminating PSTN switch, it sends a 200 OK INVITE. The TPS retransmits the 200 OK INVITE until it receives an ACK from the OPS. At this point, the call is in the talking state.

7. If the called party in the TPS goes on-hook first, the TPS can receive an ISUP SUS message. Since this call does not use encapsulated ISUP, there is no means to signal that information to the OPS, so the TPS takes no action on the SIP side of the call until it receives an ISUP REL from the terminating PSTN switch.
8. When the TPS receives an ISUP REL from the terminating PSTN switch, it formats and sends a SIP BYE to the OPS, which responds with a 200 OK BYE. The call resources are then released.

□

Detailed Call Scenarios - SIP Base

This section closely examines how SIP calls are processed within the *5ESS*® switch.

There are several different switching module processor (SMP) processes involved in the call. These processes include:

- **ISUP Terminal Process (ISUP TP):** Formulates and interprets ISUP messages. The switching module that hosts the ISUP terminal process also hosts the ISUP trunk member that has been assigned for the call.
- **SIP Terminal Process (SIP TP):** Formulates and interprets SIP messages in a compact, internal format. The SM-2000 that hosts this process is called the SIP call processing switching module.
- **Bearer Terminal Process (BRR TP):** Assigns and controls the OFI for the call. The switching module that hosts the bearer terminal process also hosts the packet group that has been assigned for the call.

These processes can be provided by the same switching module, three different switching modules, or a combination of two switching modules.

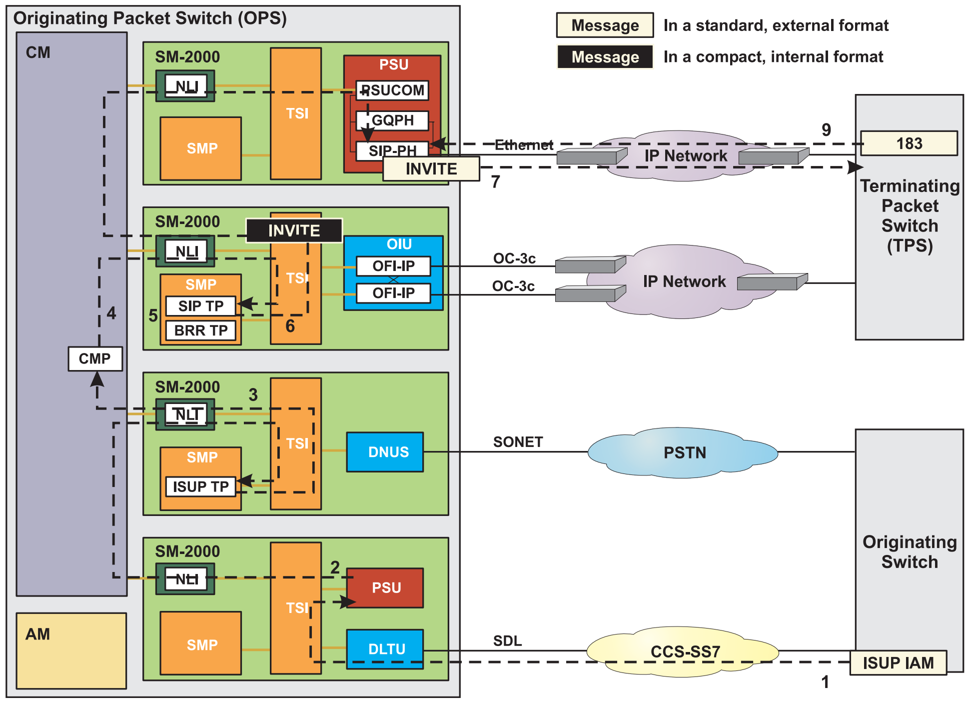
The following detailed called scenarios are discussed:

- Call Setup at Originating Packet Switch
- Call Setup at Terminating Packet Switch
- Call Tear Down at Originating Packet Switch
- Call Tear Down at Terminating Packet Switch

Note: These call scenarios use PSU-SS7 signaling.

Call Setup at Originating Packet Switch (ISUP to SIP)

This section takes a closer look at the processing and messaging performed by the OPS when acting as a gateway between the PSTN and an IP network.

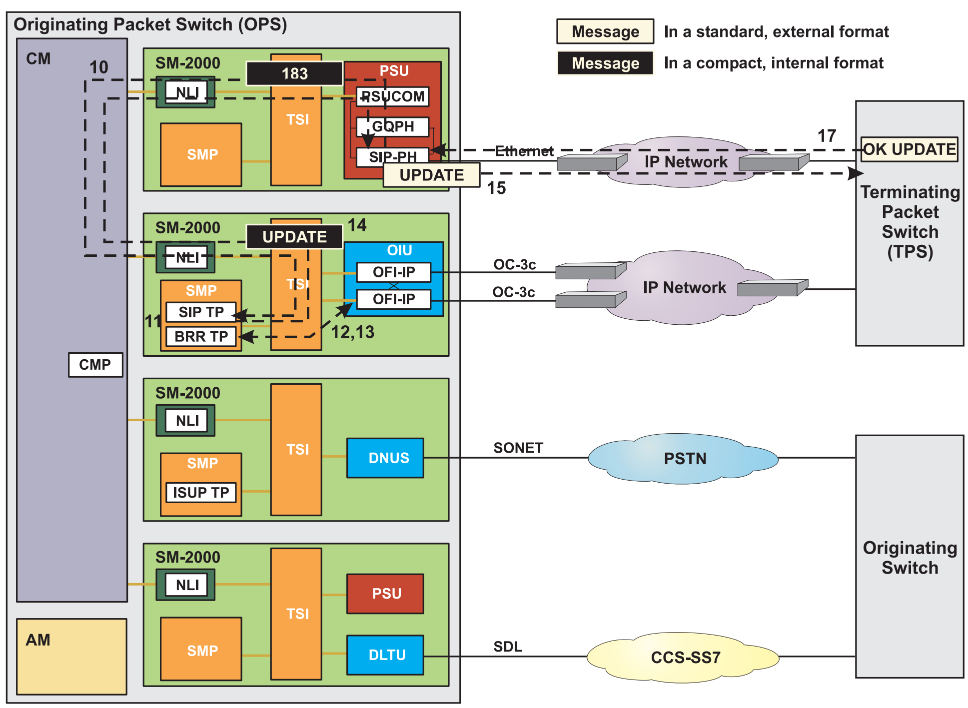


1. An ISUP IAM message arrives at the OPS on a signaling data link (SDL). The message is sent from the trunk peripheral terminating the SDL to the ST PH in the SS7 GSM via nailed up timeslots.
2. The ST PH in the PSU2 examine the message and forward it to the ISUP terminal process.
3. The ISUP terminal process performs digit analysis to determine if the call should be routed to a line or a trunk. Digit analysis determines that the call should be routed to a SIP packet group and requests that the CMP select a SIP packet group for the call.
4. The CMP:
   - determines the call is routed to a SIP packet group,
   - selects a switching module for the SIP terminal process, and
   - selects a switching module for the bearer terminal process.

   The determined information is then forwarded to the SIP terminal process and then the bearer terminal process.

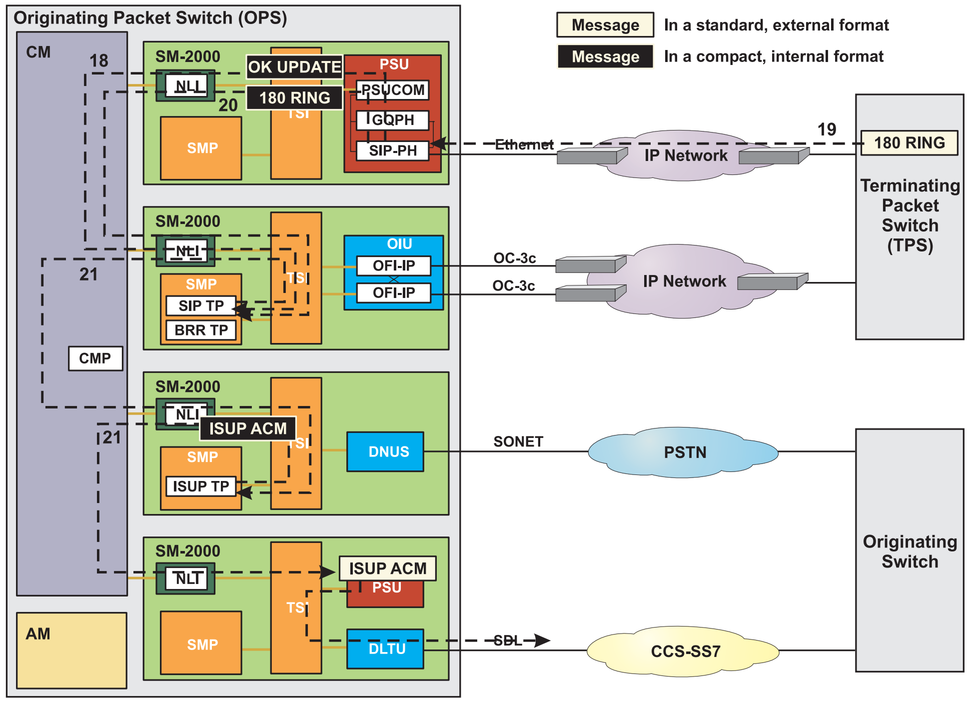
5. The bearer terminal process selects an OFI and a port for the call. The corresponding IP address and port number are forwarded to the SIP terminal process.
6. The SIP terminal process develops an INVITE message. This message is in a compact, internal SIP format. An INVITE message requests that the TPS allocate resources for the call. This message includes:
   - IP address and port number for the OPS OFI-IP
   - Encapsulated ISUP IAM message
   - A description of the requested session.

   The formulated message is forwarded to the SIP PH in the SIP GSM. Additionally, an INVITE transaction timer is started.
7. The SIP PH expands the INVITE message to standard, external SIP format, adds the appropriate SCTP and IP headers, and forwards it to TPS via the IP signaling network.
8. The OPS waits for the TPS to respond.
9. The SIP PH receives a 183 SESSION PROGRESS message from the TPS. The message includes the IP address and port number of the TPS bearer resource.



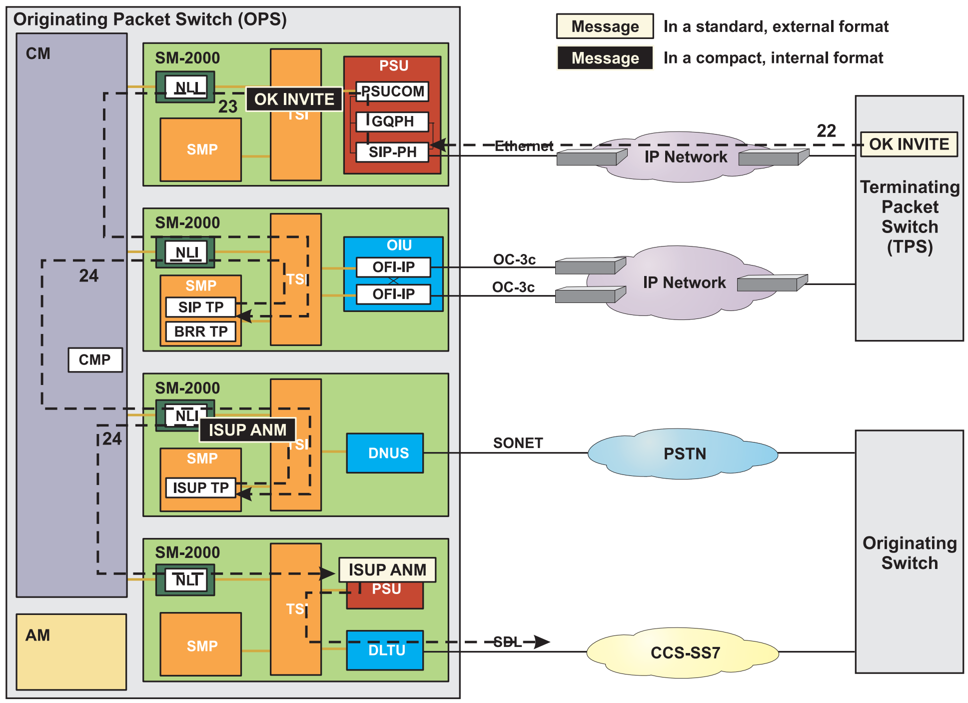
10. The SIP PH strips the SCTP and IP headers, compacts the message to the internal SIP format, and forwards the message to the SIP terminal process.
11. The SIP terminal process forwards the IP address and port number of the TPS bearer resource to the bearer terminal process.
12. The bearer terminal process forwards the IP address and port number of the TPS bearer resource to the selected OFI.
13. The selected OFI acknowledges that it received the information and opens the selected port.
14. The SIP terminal process formulates an UPDATE message and forwards it to the SIP PH in the SIP GSM. This message is in a compact, internal SIP format. The UPDATE message notifies the TPS that the bearer path is now available.
15. The SIP PH expands the UPDATE message to standard, external SIP format, adds the appropriate SCTP and IP headers, and forwards it to TPS via the IP signaling network.
16. The OPS waits for the TPS to respond.

17. The SIP PH receives a 200 OK (UPDATE) message from the TPS. This message acknowledges that the TPS received the UPDATE message.



18. The SIP PH strips the SCTP and IP headers, compacts the message to the internal SIP format, and forwards the message to the SIP terminal process.
19. The SIP PH receives a 180 RINGING message from the TPS. The message informs the OPS that the called party is available and that ringing has been applied.
20. The SIP PH strips the SCTP and IP headers, compacts the message to the internal SIP format, and forwards the message to the SIP terminal process. The INVITE transaction timer is stopped in the SIP terminal process.

21. The SIP terminal process passes notification of ringing to the ISUP terminal process. An ISUP ACM message is formulated and forwarded to the ST PH in the SS7 GSM. Finally, the message is sent to the originating switch.



22. The SIP PH receives a 200 OK (INVITE) message from the TPS. The message informs the OPS that the called party has gone off-hook.

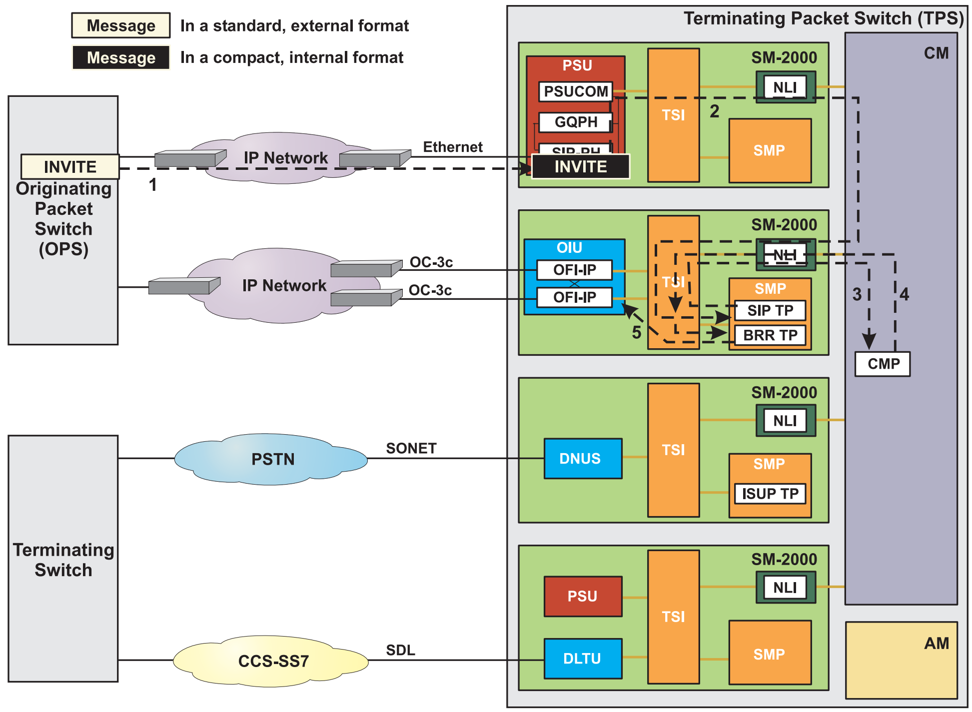
23. The SIP PH strips the SCTP and IP headers, compacts the message to the internal SIP format, and forwards the message to the SIP terminal process.

24. The SIP terminal process passes notification to the ISUP terminal process. An ISUP ANM message is formulated and forwarded to the ST PH in the SS7 GSM. Finally, the message is sent to the originating switch.

25. The call path through the switch is established and remains available until the call is terminated.

Call Setup at Terminating Packet Switch (SIP to ISUP)

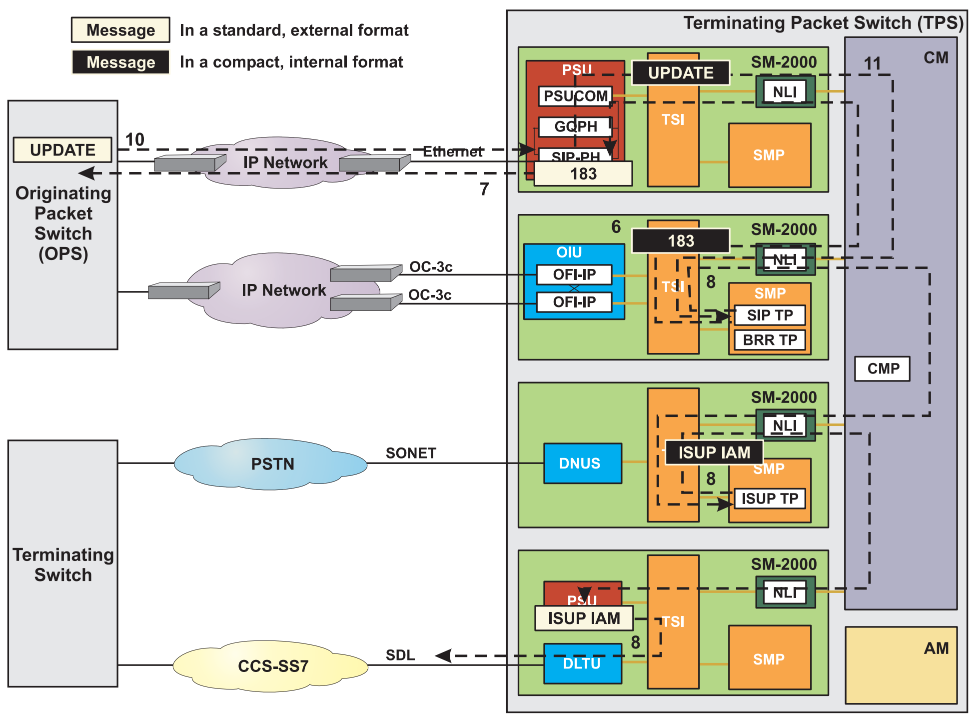
This section takes a closer look at the processing and messaging performed by the TPS when acting as a gateway between an IP network and the PSTN.



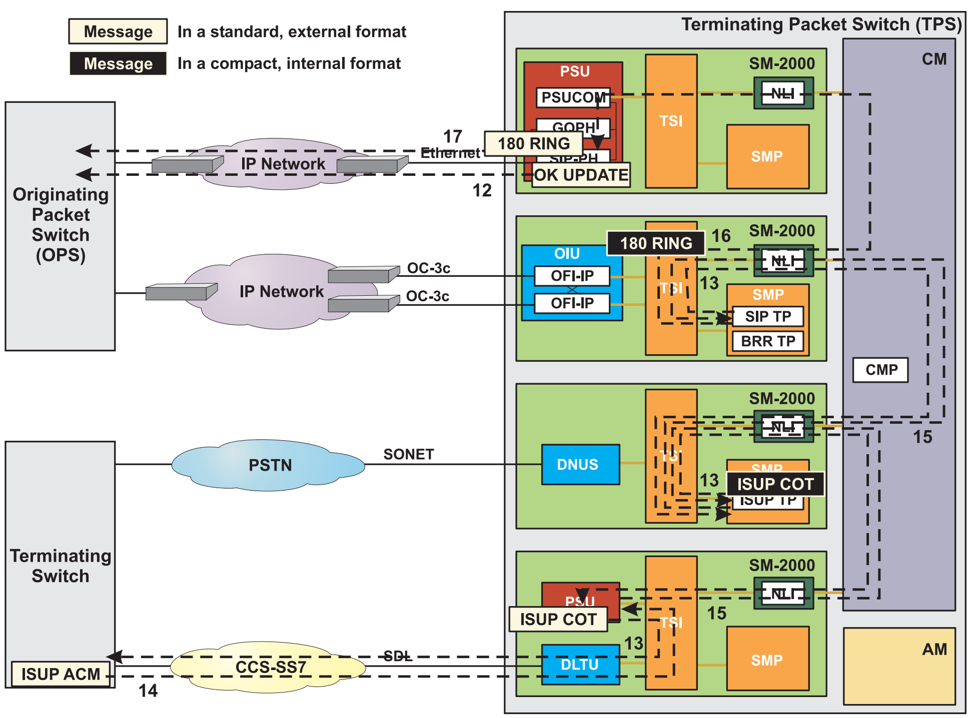
1. An INVITE message arrives at the SIP PH in the SIP GSM. This message includes:
   - IP address of OPS signaling point
   - IP address of TPS signaling point
   - IP address and port number for the OPS bearer resource
   - Encapsulated ISUP IAM message
   - A description of the requested session.
2. The SIP PH strips the SCTP and IP headers and compacts the message to the internal SIP format. The SIP PH then selects a switching module for the SIP terminal process and forwards the message to it.

3. The SIP terminal process performs digit analysis to determine if the call should be routed to a line or a trunk. The SIP terminal process determines the call should be routed to an ISUP trunk. Additionally, the SIP terminal process determines that the call from the OPS must arrive at a bearer resource so it requests that the CMP select a bearer resource for the call.
4. The CMP:
   - determines that the call arrives at a SIP packet group,
   - selects a switching module for the bearer terminal process.

   The determined information as well as the IP address and port number of the OPS bearer resource are then forwarded to the selected IP bearer switching module.
5. The bearer terminal process selects an OFI and a port for the call. The bearer terminal process request that the OFI open the selected port for this transaction. Finally, the IP address and port number of the selected resource are forwarded to the SIP terminal process.

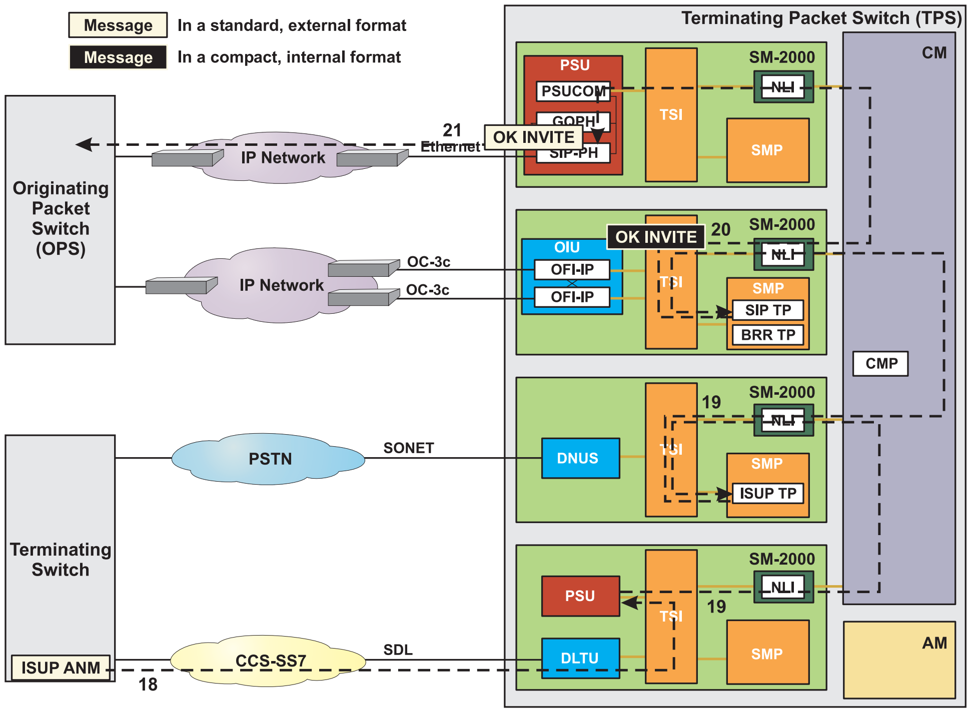


6. The SIP terminal process develops a 183 SESSION PROGRESS message. This message is in a compact, internal SIP format. This message contains the IP address and port for the terminating bearer resource. The SIP message is forwarded to the SIP PH in the SIP GSM.
7. The SIP PH expands the 183 SESSION PROGRESS message to standard, external SIP format, adds the appropriate SCTP and IP headers, and forwards it to OPS via the IP signaling network.
8. The SIP terminal process passes notification to the ISUP terminal process. An ISUP IAM message is formulated and forwarded to the ST PH in the SS7 GSM. Finally, the message is sent to the terminating switch.
9. The TPS waits for the OPS to respond.
10. The SIP PH receives an UPDATE message from the OPS. The message informs the TPS that the OPS has opened the upstream port and that the bearer path is now available.
11. The SIP PH strips the SCTP and IP headers, compacts the message to the internal SIP format, and forwards the message to the SIP terminal process.



12. The SIP PH builds the 200 OK (UPDATE) message in standard, external SIP format, adds the appropriate SCTP and IP headers, and forwards it to OPS via the IP signaling network.
13. The SIP terminal process passes the updated call state to the ISUP terminal process. An ISUP COT message is formulated and forwarded to the ST PH in the SS7 GSM. Finally, the message is sent to the terminating switch.
14. An ISUP ACM message arrives at the TPS on a signaling data link (SDL). The message is sent from the trunk peripheral terminating the SDL to the ST PH in the SS7 GSM via nailed up timeslots.
15. The ST PH examine the message and forward it to the ISUP terminal process. The message is then forwarded to the SIP terminal process.
16. The SIP terminal process develops a 180 RINGING message. This message is in a compact, internal SIP format. The SIP message is forwarded to the SIP PH in the SIP GSM.

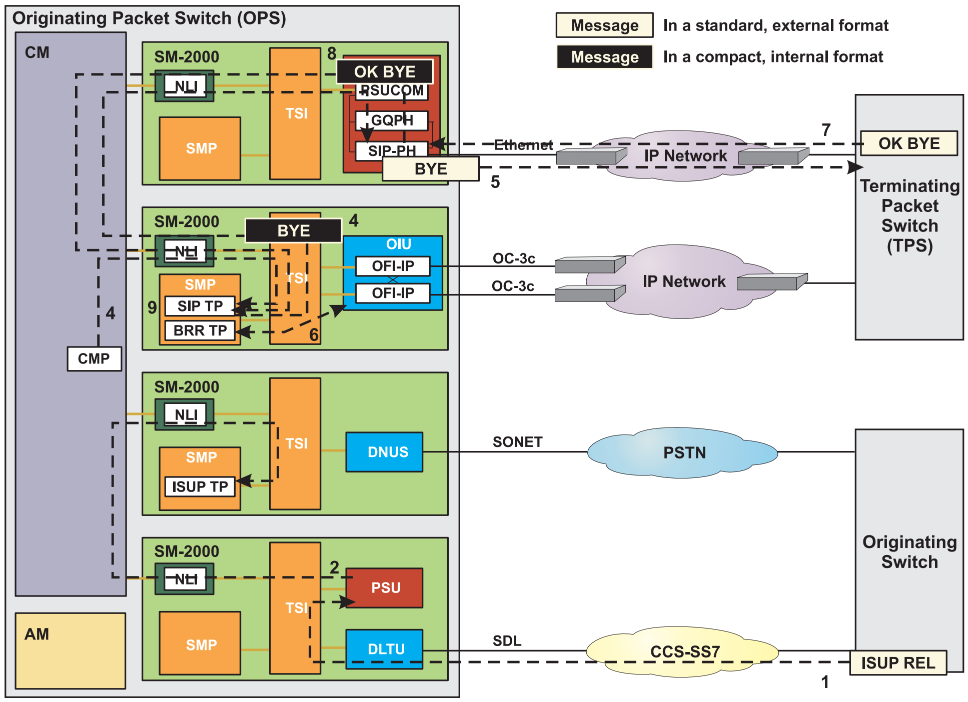
17. The SIP PH expands the 180 RINGING message to standard, external SIP format, adds the appropriate SCTP and IP headers, and forwards it to OPS via the IP signaling network.



18. An ISUP ANM message arrives at the TPS on a signaling data link (SDL). The message is sent from the trunk peripheral terminating the SDL to the ST PH in the SS7 GSM via nailed up timeslots.
19. The ST PH examine the message and forward it to the ISUP terminal process. The message is then forwarded to the SIP terminal process.
20. The SIP terminal process develops a 200 OK (INVITE) message. This message is in a compact, internal SIP format. The SIP message is forwarded to the SIP PH in the SIP GSM.
21. The SIP PH expands the 200 OK (INVITE) message to standard, external SIP format, adds the appropriate SCTP and IP headers, and forwards it to OPS via the IP signaling network.
22. The call path through the switch is established and remains available until the call is terminated.

Call Tear Down at Originating Packet Switch (ISUP to SIP)

This section takes a closer look at the processing and messaging completed by the OPS when acting as a gateway between the PSTN and an IP network .

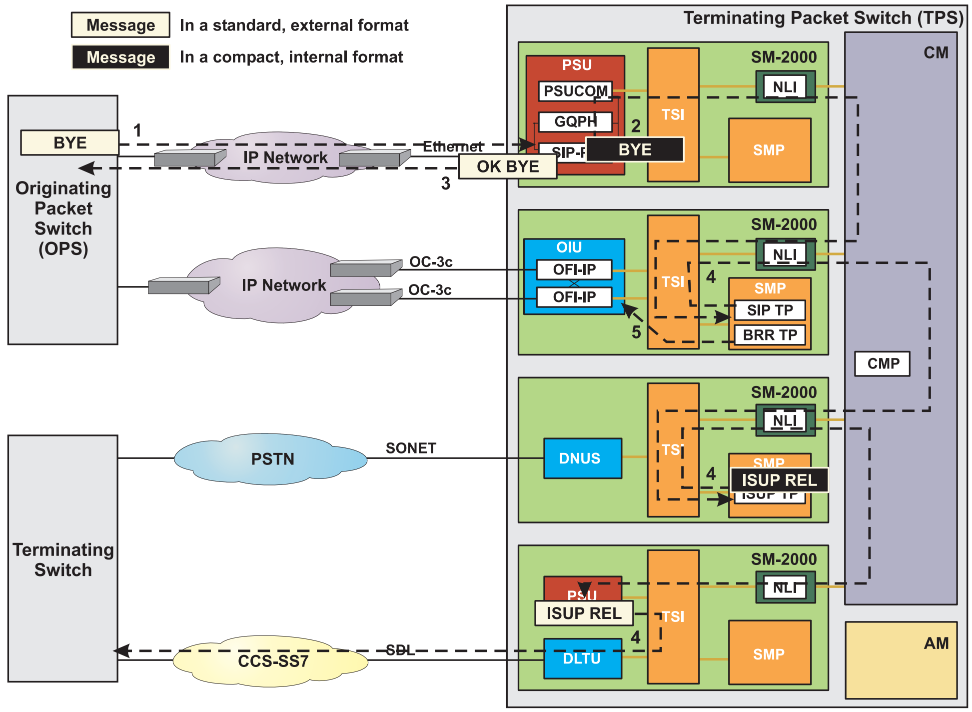


1. An ISUP REL message arrives at the OPS on a signaling data link (SDL). The message is sent from the trunk peripheral terminating the SDL to the ST PH in the SS7 GSM via nailed up timeslots.
2. The ST PH examine the message and forward it to the ISUP terminal process. The message is then forwarded to the SIP terminal process.
3. Peripheral control in the IP bearer switching module detects discontinuity and informs the SIP terminal process of the change of state.
4. The SIP terminal process formulates a BYE message and forwards it to the SIP PH in the SIP GSM. This message is in a compact, internal SIP format. This message informs the TPS to tear down the call.
5. The SIP PH expands the BYE message to standard, external SIP format, adds the appropriate SCTP and IP headers, and forwards it to TPS via the IP signaling network.

6. The bearer terminal process takes steps to tear down the call by closing the port on the OFI and releasing used timeslots.
7. The SIP PH receives a 200 OK (BYE)message from the TPS. The message informs the OPS that it received the BYE message.
8. The SIP PH strips the SCTP and IP headers, compacts the message to the internal SIP format, and forwards the message to the SIP terminal process.
9. The SIP terminal process terminates any remaining processes dedicated to the call.
10. Call flow is complete

Call Tear Down at Terminating Packet Switch (SIP to ISUP)

This section takes a closer look at the processing and messaging completed by the TPS when acting as a gateway between the PSTN and an IP network .



1. The SIP PH receives a BYE message from the OPS. The message informs the TPS the calling party went on-hook..
2. The SIP PH strips the SCTP and IP headers, compacts the message to the internal SIP format, and forwards the message to the SIP terminal process.
3. The SIP PH builds the 200 OK (BYE) message in standard, external SIP format, adds the appropriate SCTP and IP headers, and forwards it to OPS via the IP signaling network.
4. The SIP terminal process passes the updated call state to the ISUP terminal process. An ISUP REL message is formulated and forwarded to the ST PH in the SS7 GSM. Finally, the message is sent to the terminating switch.
5. The SIP terminal process passes the updated call state to the bearer terminal process. The bearer terminal process takes steps to tear down the call by closing the port on the OFI and releasing used timeslots.
6. Call flow is complete

SIP Message Examples

This section includes example SIP messages to illustrate their format and content. Ethernet, IP, and SCTP headers were removed to focus on the SIP message and any encapsulated ISUP and SDP messages.

Figure 3-8 SIP INVITE

```
INVITE
Session Initiation Protocol
    Request line: INVITE sip:2208000@10.11.3.25;transport=sctp;user=phone SIP/
2.0
        Method: INVITE
    Message Header
        t: <sip:2208000@10.11.3.25;transport=sctp;user=phone>
        f: <sip:+18152207000@10.11.3.1;transport=sctp;user=phone>;tag=0-01-012-
0000f-17b5
        i: 00003047-00020039-0002307b@LABM0
        CSeq: 1 INVITE
        v: SIP/2.0/SCTP 10.11.3.1:1;branch=z9hG4bK012001d.00000001
        m: <10.11.3.1:1>
        Max-Forwards: 20
        Accept: application/SDP, application/ISUP, multipart/mixed
        MIME-Version: 1.0
        c: multipart/mixed;boundary=unique-boundary-1
        Allow: INVITE, ACK, BYE, CANCEL, UPDATE, INFO, OPTIONS
        Require: precondition
        l: 477
Session Description Protocol
    Session Description Protocol Version (v): 0
    Owner/Creator, Session Id (o): - 0 0 IN IP4 192.168.3.5
        Owner Username: -
        Session ID: 0
        Session Version: 0
        Owner Network Type: IN
        Owner Address Type: IP4
        Owner Address: 192.168.3.5
    Session Name (s): 0
    Connection Information (c): IN IP4 192.168.3.5
        Connection Network Type: IN
        Connection Address Type: IP4
        Connection Address: 192.168.3.5
    Time Description, active time (t): 0 0
        Session Start Time: 0
        Session Start Time: 0
    Media Description, name and address (m): audio 49180 RTP/AVP 0
        Media Type: audio
        Media Port: 49180
        Media Proto: RTP/AVP
        Media Format: 0
    Bandwidth Information (b): AS:64
        Bandwidth Modifier: AS
```

```
            Bandwidth Value: 64
        Media Attribute (a): ptime:20
            Media Attribute Fieldname: ptime
            Media Attribute Value: 20
        Media Attribute (a): rtpmap:0 PCMU/8000
            Media Attribute Fieldname: rtpmap
            Media Attribute Value: 0 PCMU/8000
        Media Attribute (a): curr:qos local none
            Media Attribute Fieldname: curr
            Media Attribute Value: qos local none
        Media Attribute (a): curr:qos remote none
            Media Attribute Fieldname: curr
            Media Attribute Value: qos remote none
        Media Attribute (a): des:qos mandatory local sendrecv
            Media Attribute Fieldname: des
            Media Attribute Value: qos mandatory local sendrecv
        Media Attribute (a): des:qos mandatory remote sendrecv
            Media Attribute Fieldname: des
            Media Attribute Value: qos mandatory remote sendrecv
    ISDN User Part
        Message type: IAM  : Initial Address Message (1)
        Nature of Connection Indicators: 0x0
            Mandatory Parameter: 6 (Nature of connection indicators)
            .... ..00 = Satellite Indicator: No Satellite circuit in connection
    (0x00)
            .... 00.. = Continuity Check Indicator: Continuity check not required
    (0x00)
            ...0 .... = Echo Control Device Indicator: Echo control device not
    included
        Forward Call Indicators: 0x2000
            Mandatory Parameter: 7 (Forward call indicators)
            .... ...0 .... .... = National/international call indicator: Call to be
    treated as national call
            .... .00. .... .... = End-to-end method indicator: No End-to-end method
    available (only link-by-link method available) (0x0000)
            .... 0... .... .... = Interworking indicator: no interworking
    encountered (No.7 signalling all the way)
            ...0 .... .... .... = End-to-end information indicator: no end-to-end
    information available
            ..1. .... .... .... = ISDN user part indicator: ISDN user part used all
    the way
            00.. .... .... .... = ISDN user part preference indicator: ISDN user
    part prefered all the way (0x0000)
            .... .... .... ...0 = ISDN access indicator: originating access non-ISDN
            .... .... .... .00. = SCCP method indicator: No indication (0x0000)
        Calling Party's category: 0xa (ordinary calling subscriber)
            Mandatory Parameter: 9 (Calling party's category)
            Calling Party's category: ordinary calling subscriber (0x0a)
        User Service Information
            coding std. = 0 (CCITT standardized in this recommendation)
            info. transfer capability = 0x0 (speech)
            transfer mode = 0 (circuit mode)
            info. trans. rate (d<->o) = 0x10 (64 kbit/s (for analog circuit))
            structure = default (8kHz integrity)
            config. = default (point-to-point)
            establishment = default (demand)
```

```
        symmetry = default (bidirectional symmetric)
        user info. layer 1 protocol id = 0x2
        (u-law speech)
    Called Party Number: 2208000
        Mandatory Parameter: 4 (Called party number)
        Pointer to Parameter: 6
        Parameter length: 6
        1... .... = Odd/even indicator: odd number of address signals
        .000 0001 = Nature of address indicator: subscriber number (national
use) (1)
        0... .... = INN indicator: routing to internal network number allowed
        .001 .... = Numbering plan indicator: ISDN (Telephony) numbering plan
(1)
        Called Party Number: 2208000
            .... 0010 = Address signal digit: 2 (2)
            0010 .... = Address signal digit: 2 (2)
            .... 0000 = Address signal digit: 0 (0)
            1000 .... = Address signal digit: 8 (8)
            .... 0000 = Address signal digit: 0 (0)
            0000 .... = Address signal digit: 0 (0)
            .... 0000 = Address signal digit: 0 (0)
    Pointer to start of optional part: 12
    Calling Party Number: 8152207000
        Optional Parameter: 10 (Calling party number)
        Parameter length: 7
        0... .... = Odd/even indicator: even number of address signals
        .000 0011 = Nature of address indicator: national (significant) number
(3)
        0... .... = NI indicator: complete
        .001 .... = Numbering plan indicator: ISDN (Telephony) numbering plan
(1)
        .... 00.. = Address presentation restricted indicator: presentation
allowed (0)
        .... ..11 = Screening indicator: network provided (3)
        Calling Party Number: 8152207000
            .... 1000 = Address signal digit: 8 (8)
            0001 .... = Address signal digit: 1 (1)
            .... 0101 = Address signal digit: 5 (5)
            0010 .... = Address signal digit: 2 (2)
            .... 0010 = Address signal digit: 2 (2)
            0000 .... = Address signal digit: 0 (0)
            .... 0111 = Address signal digit: 7 (7)
            0000 .... = Address signal digit: 0 (0)
            .... 0000 = Address signal digit: 0 (0)
            0000 .... = Address signal digit: 0 (0)
    Parameter Type unknown/reserved (7 Bytes)
        Optional Parameter: 235 (unknown)
        Parameter length: 7
    Parameter Type unknown/reserved (3 Bytes)
        Optional Parameter: 196 (unknown)
        Parameter length: 3
    End of optional parameters (0)
```

Figure 3-9 SIP 183 SESSION PROGRESS

```
183 SESSION PROGRESS
Session Initiation Protocol
    Status line: SIP/2.0 183 Session Progress
        Status-Code: 183
    Message Header
        t: <sip:2208000@10.11.3.25;transport=sctp;user=phone>;tag=0-04-012-40007
-17b7
        f: <sip:+18152207000@10.11.3.1;transport=sctp;user=phone>;tag=0-01-012-
0000f-17b5
        i: 00003047-00020039-0002307b@LABMO
        CSeq: 1 INVITE
        v: SIP/2.0/SCTP 10.11.3.1:1;branch=z9hG4bK012001d.00000001
        MIME-Version: 1.0
        c: application/SDP
        Allow: INVITE, ACK, BYE, CANCEL, UPDATE, INFO, OPTIONS
        Require: precondition
        l: 283
Session Description Protocol
    Session Description Protocol Version (v): 0
    Owner/Creator, Session Id (o): - 0 0 IN IP4 192.168.3.5
        Owner Username: -
        Session ID: 0
        Session Version: 0
        Owner Network Type: IN
        Owner Address Type: IP4
        Owner Address: 192.168.3.5
    Session Name (s): 0
    Connection Information (c): IN IP4 192.168.10.20
        Connection Network Type: IN
        Connection Address Type: IP4
        Connection Address: 192.168.10.20
    Time Description, active time (t): 0 0
        Session Start Time: 0
        Session Start Time: 0
    Media Description, name and address (m): audio 50620 RTP/AVP 0
        Media Type: audio
        Media Port: 50620
        Media Proto: RTP/AVP
        Media Format: 0
    Bandwidth Information (b): AS:64
        Bandwidth Modifier: AS
        Bandwidth Value: 64
    Media Attribute (a): ptime:20
        Media Attribute Fieldname: ptime
        Media Attribute Value: 20
    Media Attribute (a): rtpmap:0 PCMU/8000
        Media Attribute Fieldname: rtpmap
        Media Attribute Value: 0 PCMU/8000
    Media Attribute (a): curr:qos local none
        Media Attribute Fieldname: curr
        Media Attribute Value: qos local none
    Media Attribute (a): curr:qos remote none
        Media Attribute Fieldname: curr
```

```
        Media Attribute Value: qos remote none
    Media Attribute (a): des:qos mandatory local sendrecv
        Media Attribute Fieldname: des
        Media Attribute Value: qos mandatory local sendrecv
    Media Attribute (a): des:qos mandatory remote sendrecv
        Media Attribute Fieldname: des
        Media Attribute Value: qos mandatory remote sendrecv
    Media Attribute (a): conf:qos remote sendrecv
        Media Attribute Fieldname: conf
        Media Attribute Value: qos remote sendrecv
```

Figure 3-10 SIP UPDATE

```
UPDATE
Session Initiation Protocol
    Request line: UPDATE sip:2208000@10.11.3.25;transport=sctp;user=phone SIP/
2.0
        Method: UPDATE
    Message Header
        t: <sip:2208000@10.11.3.25;transport=sctp;user=phone>;tag=0-04-012-40007
-17b7
        f: <sip:+18152207000@10.11.3.1;transport=sctp;user=phone>;tag=0-01-012-
0000f-17b5
        i: 00003047-00020039-0002307b@LABM0
        CSeq: 2 UPDATE
        v: SIP/2.0/SCTP 10.11.3.1:1;branch=z9hG4bK012001d.00000002
        m: <10.11.3.1:1>
        Max-Forwards: 20
        MIME-Version: 1.0
        c: application/SDP
        l: 259
Session Description Protocol
    Session Description Protocol Version (v): 0
    Owner/Creator, Session Id (o): - 0 0 IN IP4 192.168.3.5
        Owner Username: -
        Session ID: 0
        Session Version: 0
        Owner Network Type: IN
        Owner Address Type: IP4
        Owner Address:192.168.3.5
    Session Name (s): 0
    Connection Information (c): IN IP4 192.168.3.5
        Connection Network Type: IN
        Connection Address Type: IP4
        Connection Address: 192.168.3.5
    Time Description, active time (t): 0 0
        Session Start Time: 0
        Session Start Time: 0
    Media Description, name and address (m): audio 49180 RTP/AVP 0
        Media Type: audio
        Media Port: 49180
        Media Proto: RTP/AVP
        Media Format: 0
    Bandwidth Information (b): AS:64
        Bandwidth Modifier: AS
```

```
        Bandwidth Value: 64
    Media Attribute (a): ptime:20
        Media Attribute Fieldname: ptime
        Media Attribute Value: 20
    Media Attribute (a): rtpmap:0 PCMU/8000
        Media Attribute Fieldname: rtpmap
        Media Attribute Value: 0 PCMU/8000
    Media Attribute (a): curr:qos local sendrecv
        Media Attribute Fieldname: curr
        Media Attribute Value: qos local sendrecv
    Media Attribute (a): curr:qos remote none
        Media Attribute Fieldname: curr
        Media Attribute Value: qos remote none
    Media Attribute (a): des:qos mandatory local sendrecv
        Media Attribute Fieldname: des
        Media Attribute Value: qos mandatory local sendrecv
    Media Attribute (a): des:qos mandatory remote sendrecv
        Media Attribute Fieldname: des
        Media Attribute Value: qos mandatory remote sendrecv
```

Figure 3-11 SIP 200 OK (UPDATE)

```
200 OK (UPDATE)
Session Initiation Protocol
    Status line: SIP/2.0 200 OK
        Status-Code: 200
    Message Header
        t: <sip:2208000@10.11.3.25;transport=sctp;user=phone>;tag=0-04-012-40007
-17b7
        f: <sip:+18152207000@10.11.3.1;transport=sctp;user=phone>;tag=0-01-012-
0000f-17b5
        i: 00003047-00020039-0002307b@LABM0
        CSeq: 2 UPDATE
        v: SIP/2.0/SCTP 10.11.3.1:1;branch=z9hG4bK012001d.00000002
        m: <10.11.3.25:1>
        MIME-Version: 1.0
        c: application/SDP
        l: 263
Session Description Protocol
    Session Description Protocol Version (v): 0
    Owner/Creator, Session Id (o): - 0 0 IN IP4 192.168.3.5
        Owner Username: -
        Session ID: 0
        Session Version: 0
        Owner Network Type: IN
        Owner Address Type: IP4
        Owner Address: 192.168.3.5
    Session Name (s): 0
    Connection Information (c): IN IP4 192.168.3.5
        Connection Network Type: IN
        Connection Address Type: IP4
        Connection Address: 192.168.3.5
    Time Description, active time (t): 0 0
        Session Start Time: 0
        Session Start Time: 0
```

```
    Media Description, name and address (m): audio 50620 RTP/AVP 0
        Media Type: audio
        Media Port: 50620
        Media Proto: RTP/AVP
        Media Format: 0
    Bandwidth Information (b): AS:64
        Bandwidth Modifier: AS
        Bandwidth Value: 64
    Media Attribute (a): ptime:20
        Media Attribute Fieldname: ptime
        Media Attribute Value: 20
    Media Attribute (a): rtpmap:0 PCMU/8000
        Media Attribute Fieldname: rtpmap
        Media Attribute Value: 0 PCMU/8000
    Media Attribute (a): curr:qos local sendrecv
        Media Attribute Fieldname: curr
        Media Attribute Value: qos local sendrecv
    Media Attribute (a): curr:qos remote sendrecv
        Media Attribute Fieldname: curr
        Media Attribute Value: qos remote sendrecv
    Media Attribute (a): des:qos mandatory local sendrecv
        Media Attribute Fieldname: des
        Media Attribute Value: qos mandatory local sendrecv
    Media Attribute (a): des:qos mandatory remote sendrecv
        Media Attribute Fieldname: des
        Media Attribute Value: qos mandatory remote sendrecv
```

Figure 3-12 SIP 180 RINGING

```
180 RINGING
Session Initiation Protocol
    Status line: SIP/2.0 180 Ringing
        Status-Code: 180
    Message Header
        t: <sip:2208000@10.11.3.25;transport=sctp;user=phone>;tag=0-04-012-40007
-17b7
        f: <sip:+18152207000@10.11.3.1;transport=sctp;user=phone>;tag=0-01-012-
0000f-17b5
        i: 00003047-00020039-0002307b@LABM0
        CSeq: 1 INVITE
        v: SIP/2.0/SCTP 10.11.3.1:1;branch=z9hG4bK012001d.00000001
        MIME-Version: 1.0
        c: application/ISUP; version=ansi00
        Content-Disposition: signal; handling=required
        l: 4
ISDN User Part
    Message type: ACM  : Address complete message (6)
    Backward Call Indicators: 0x1424
        Mandatory Parameter: 17 (Backward call indicators)
        .... ..00 .... .... = Charge indicator: No indication (0x0000)
        .... 01.. .... .... = Called party’s status indicator: Subscriber free
(0x0001)
        ..01 .... .... .... = Called party’s category indicator: Ordinary
subscriber (0x0001)
        00.. .... .... .... = End-to-end method indicator: No End-to-end method
```

```
available (only link-by-link method available) (0x0000)
        .... .... .... ...0 = Interworking indicator: no interworking
encountered (No.7 signalling all the way)
        .... .... .... ..0. = End-to-end information indicator: no end-to-end
information available
        .... .... .... .1.. = ISDN user part indicator: ISDN user part used all
the way
        .... .... .... 0... = Holding indicator: holding not requested
        .... .... ...0 .... = ISDN access indicator: terminating access non-ISDN
        .... .... ..1. .... = Echo Control Device Indicator: Echo control device
 included
        .... .... 00.. .... = SCCP method indicator: No indication (0x0000)
    No optional parameter present (Pointer: 0)
```

Figure 3-13 SIP 200 OK (INVITE)

```
200 OK (INVITE)
Session Initiation Protocol
    Status line: SIP/2.0 200 OK
        Status-Code: 200
    Message Header
        t: <sip:2208000@10.11.3.25;transport=sctp;user=phone>;tag=0-04-012-40007
-17b7
        f: <sip:+18152207000@10.11.3.1;transport=sctp;user=phone>;tag=0-01-012-
0000f-17b5
        i: 00003047-00020039-0002307b@LABMO
        CSeq: 1 INVITE
        v: SIP/2.0/SCTP 10.11.3.1:1;branch=z9hG4bK012001d.00000001
        m: <10.11.3.25:1>
        MIME-Version: 1.0
        c: application/ISUP; version=ansi00
        Content-Disposition: signal; handling=required
        Allow: INVITE, ACK, BYE, CANCEL, UPDATE, INFO, OPTIONS
        l: 2
ISDN User Part
    Message type: ANS  : Answer message (9)
    No optional parameter present (Pointer: 0)
```

Figure 3-14 SIP BYE

```
BYE
Session Initiation Protocol
    Request line: BYE sip:2208000@10.11.3.25;transport=sctp;user=phone SIP/2.0
        Method: BYE
    Message Header
        t: <sip:2208000@10.11.3.25;transport=sctp;user=phone>;tag=0-04-012-40007
-17b7
        f: <sip:+18152207000@10.11.3.1;transport=sctp;user=phone>;tag=0-01-012-
0000f-17b5
        i: 00003047-00020039-0002307b@LABMO
        CSeq: 3 BYE
        v: SIP/2.0/SCTP 10.11.3.1:1;branch=z9hG4bK012001d.00000003
        Max-Forwards: 20
        MIME-Version: 1.0
```

```
        c: application/ISUP; version=ansi00
        Content-Disposition: signal; handling=required
        l: 6
ISDN User Part
    Message type: REL  : Release message (12)
    Cause indicators, see Q.850 (2 bytes length)
        Mandatory Parameter: 18 (Cause indicators)
        Pointer to Parameter: 2
        Parameter length: 2
        Cause indicators (-> Q.850)
        Coding standard: ITU-T standardized coding
        Location: Public network serving the local user (LN)
        Cause indicator: Normal call clearing (16)
    No optional parameter present (Pointer: 0)
```

Figure 3-15 SIP 200 OK (BYE)

```
200 OK (BYE)
Session Initiation Protocol
    Status line: SIP/2.0 200 OK
        Status-Code: 200
    Message Header
        t: <sip:2208000@10.11.3.25;transport=sctp;user=phone>;tag=0-04-012-40007
-17b7
        f: <sip:+18152207000@10.11.3.1;transport=sctp;user=phone>;tag=0-01-012-
0000f-17b5
        i: 00003047-00020039-0002307b@LABM0
        CSeq: 3 BYE
        v: SIP/2.0/SCTP 10.11.3.1:1;branch=z9hG4bK012001d.00000003
        l: 0
```

□

SIP without Encapsulated ISUP - Message Flow

Figure 3-16, "SIP without Encapsulated ISUP Successful Call" (3-49) illustrates the message flow using the SIP without Encapsulated ISUP feature. These are the messages sent between network elements in order to successfully setup and tear down a call between two subscribers. This example is a PSTN to IP call, with Preconditions and an INVITE without IAM.

Figure 3-16 SIP without Encapsulated ISUP Successful Call

```
Originating                                   Terminating
      Switch     OPS                   TPS         Switch
         | --IAM-> |                    |            |
         |         | ---INVITE {SDP}--> |            |
         |         |                    |  --IAM-->  |
         |         |    183 SESSION     |            |
         |         | <-PROGRESS {SDP}-- |            |
         |         |                    |            |
         |         | ---UPDATE {SDP}--> |            |
         |         |                    |            |
         |         |  200 OK (UPDATE)   |            |
         |         | <------{SDP}------ |            |
         |         |                    | <--ACM---- |
         |         | <---180 RINGING--- |            |
         | <-ACM-- |                    |            |
         |         |                    | <--ANM---  |
         |         | <-200 OK (INVITE)- |            |
         | <-ANM-- |                    |            |
         |         | ------ ACK ------> |            |
         |                                           |
         |  <----- Talking Path Available  ------->  |
         |                                           |
         | --REL-> |                    |            |
         |         | ------ BYE ------> |            |
         |         |                    | ---REL-->  |
         |         | <--200 OK (BYE)--- |            |
         |         |                    |            |
```

SIP without Encapsulated ISUP - Detailed Call Scenario

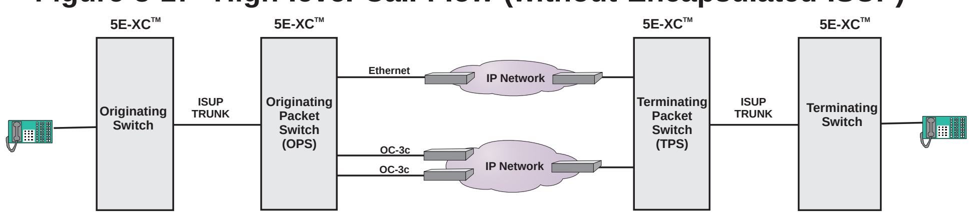
This section will provide a sample call and describes how the call is handled when SIP without Encapsulated ISUP is **on** and Encapsulated ISUP is marked **no** in RC/V. Terminology defined in the Detailed Call Scenarios - SIP Base is applicable to this section. The Detailed Call Scenarios - SIP Base section can be referenced to compare the differences in call flow.

The following scenario is described in this section:

- Public Switched Telephone Network (PSTN) to Internet Protocol (IP) call,
- using SCTP transport,
- with preconditions, and
- without Encapsulated ISUP.

A high-level view of the call scenario being described can be found in Figure 3-17, "High-level Call Flow (without Encapsulated ISUP)" (3-50). This scenario involves four different *5E-XC*™ switches. The focus of the description is SIP signaling, specifically without Encapsulated ISUP.

Figure 3-17 High-level Call Flow (without Encapsulated ISUP)



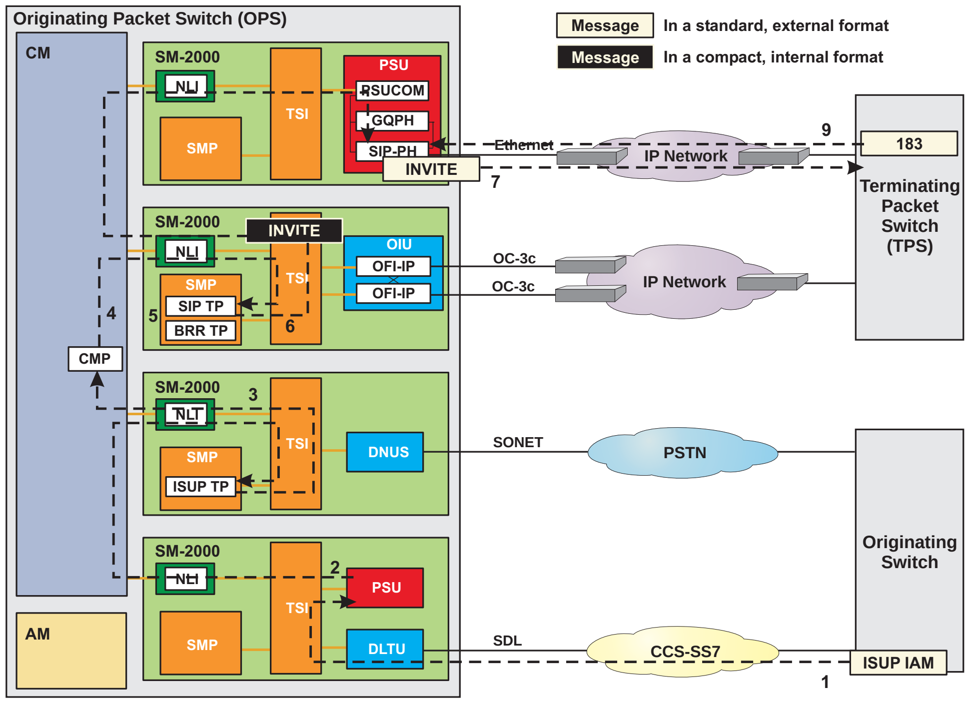
The call scenario will be divided and discussed in the following segments:

- Call Setup at OPS
- Call Setup at TPS
- Call Tear Down at OPS
- Call Tear Down at TPS

Call Setup at OPS - No ISUP

This section takes a closer look at the processing and messaging performed by the OPS when acting as a gateway between the PSTN and an IP network.

Figure 3-18 Call Setup at OPS - Steps 1-9



1. An ISUP IAM message arrives at the OPS on a signaling data link (SDL). The message is sent from the trunk peripheral terminating the SDL to the ST PH in the SS7 GSM via nailed up timeslots.
2. The ST PH in the PSU2 examines the message and forwards it to the ISUP terminal process.
3. The ISUP terminal process performs digit analysis to determine if the call should be routed to a line or a trunk. Digit analysis determines that the call should be routed to a SIP packet group and requests that the CMP select a SIP packet group for the call.
4. The CMP:
   - determines the call is routed to a SIP packet group,
   - selects a switching module for the SIP terminal process, and
   - selects a switching module for the bearer terminal process.

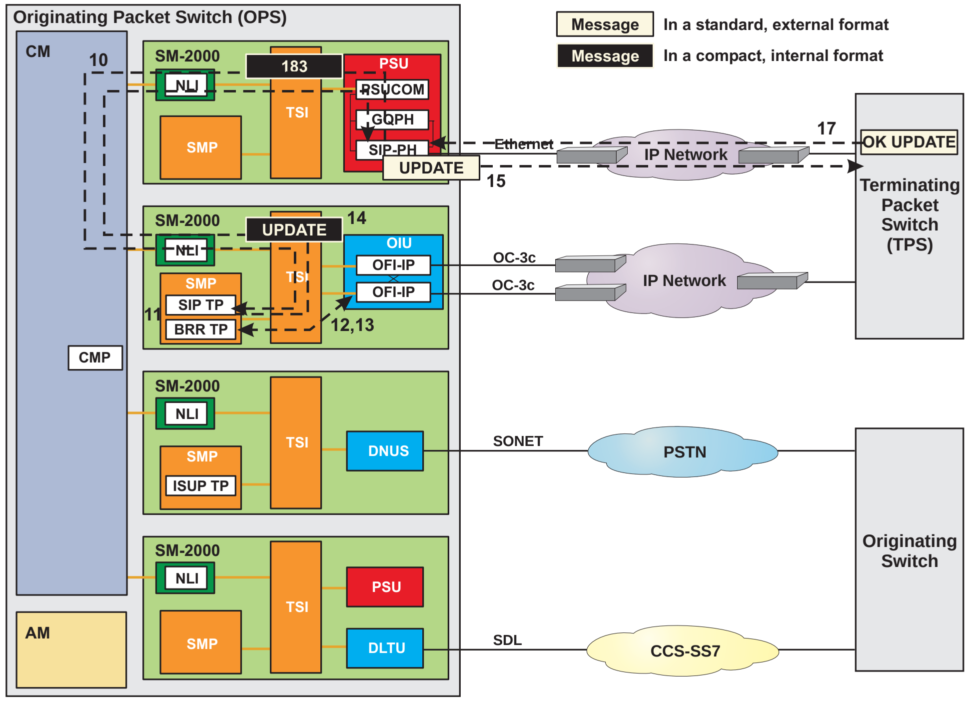
   The determined information is forwarded to the SIP terminal process and then the bearer terminal process.

5. The bearer terminal process selects an OFI and a port for the call. The corresponding IP address and port number are forwarded to the SIP terminal process.
6. The SIP terminal process creates an INVITE message. This message is in a compact, internal SIP format. During the construction of the INVITE message, the SIP terminal process checks to see if SFID 769 (SIP without Encapsulated ISUP) is **on**. If the SFID is on, the ISUP Encapsulation flag is checked (in this example ISUP Encapsulation is set to **no**). Finally, the Address Complete Message (ACM) timer is started. An INVITE message requests that the TPS allocate resources for the call. This message includes:
   - IP address and port number for the OPS OFI-IP
   - Encapsulated ISUP IAM message
   - A description of the requested session.

   The message is forwarded to the SIP PH in the SIP GSM. At this point the ISUP MIME is included. Additionally, an INVITE transaction timer is started.
7. The SIP PH expands the INVITE message to standard, external SIP format, adds the appropriate SCTP and IP headers. Prior to sending the INVITE message, a check is performed to determine whether ISUP is allowed. If not, as in this example, the ISUP MIME is discarded and the message is forwarded to the TPS via the IP signaling network.
8. The OPS waits for the TPS to respond.

9. The SIP PH receives and accepts the 183 SESSION PROGRESS message from the TPS. The message includes the IP address and port number of the TPS bearer resource.

Figure 3-19 Call Setup at OPS - Steps 10-17



10. The SIP PH strips the SCTP and IP headers, compacts the message to the internal SIP format, and forwards the message to the SIP terminal process.

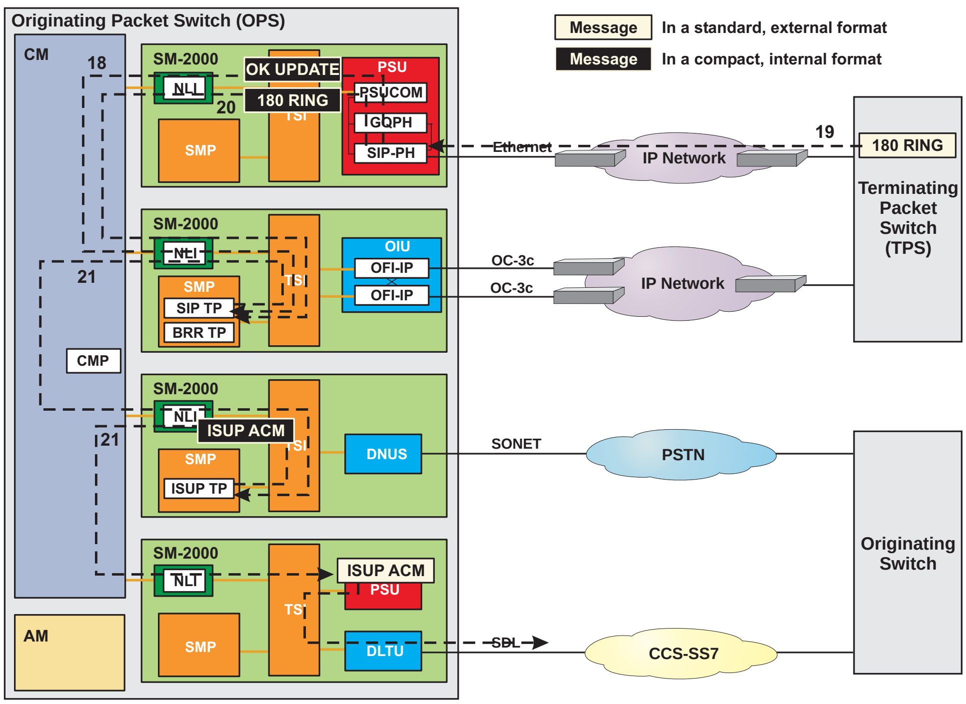
11. The SIP terminal process receives the 183 SESSION PROGRESS message and does not detect an ISUP MIME. A check is performed to verify that Encapsulated ISUP is not expected and processing continues. The SIP terminal process also forwards the IP address and port number of the TPS bearer resource to the bearer terminal process.

12. The bearer terminal process forwards the IP address and port number of the TPS bearer resource to the selected OFI.

13. The selected OFI acknowledges that it received the information and opens the selected port.

14. The SIP terminal process creates an UPDATE message and forwards it to the SIP PH in the SIP GSM. This message is in a compact, internal SIP format. The UPDATE message notifies the TPS that the bearer path is now available.
15. The SIP PH expands the UPDATE message to standard, external SIP format, adds the appropriate SCTP and IP headers, and forwards it to TPS via the IP signaling network.
16. The OPS waits for the TPS to respond.
17. The SIP PH receives and accepts the 200 OK (UPDATE) message from the TPS. This message acknowledges that the TPS received the UPDATE message.

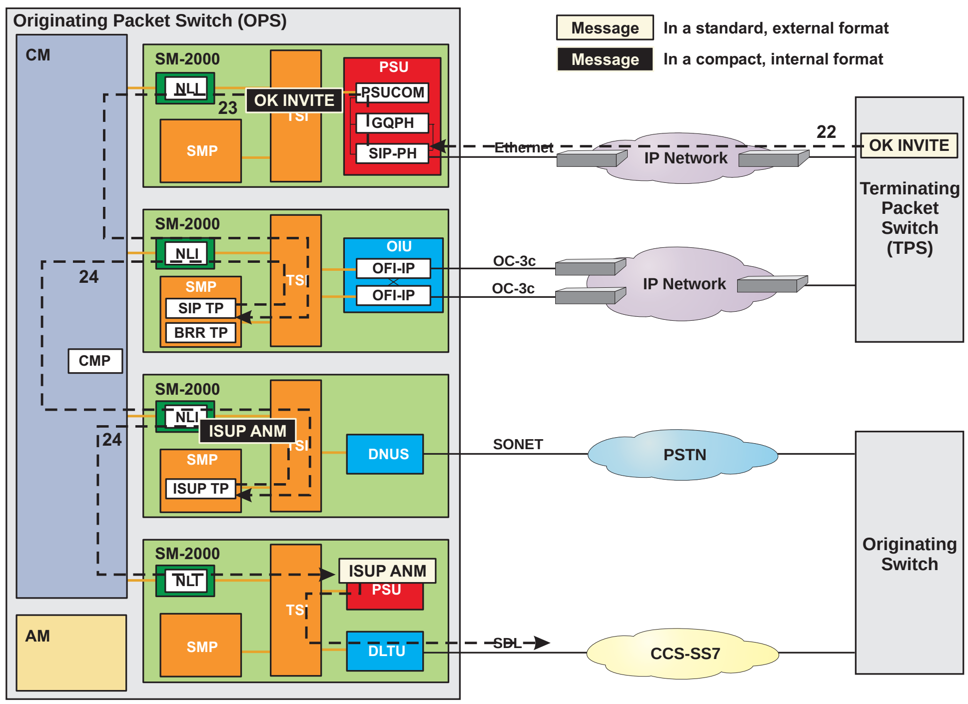
Figure 3-20 Call Setup at OPS - Steps 18-21



18. The SIP PH strips the SCTP and IP headers, compacts the message to the internal SIP format, and forwards the message to the SIP terminal process.
19. The SIP PH receives a 180 RINGING message from the TPS. The message informs the OPS that the called party is available and that ringing has been applied.

20. The SIP PH strips the SCTP and IP headers, compacts the message to the internal SIP format, and forwards the message to the SIP terminal process. The INVITE transaction timer is stopped in the SIP terminal process.
21. The SIP terminal process detects no ISUP MIME. A check is performed to verify that encapsulated ISUP is not expected, the Automatic ACM timer is stopped and processing continues. The SIP terminal process passes notification of ringing to the ISUP terminal process. An ISUP ACM message is formulated and forwarded to the ST PH in the SS7 GSM. Finally, the message is sent to the originating switch.

Figure 3-21 Call Setup at OPS - Steps 22-25



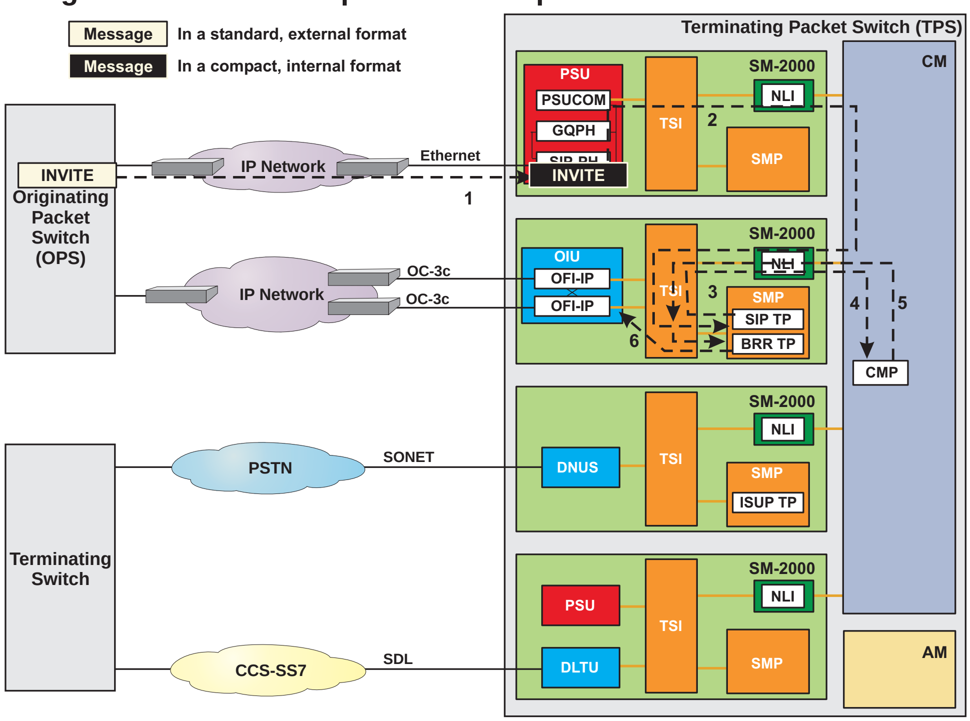
22. The SIP PH receives and accepts the 200 OK (INVITE) message from the TPS. The message informs the OPS that the called party has gone off-hook.
23. The SIP PH strips the SCTP and IP headers, compacts the message to the internal SIP format, and forwards the message to the SIP terminal process.

24. The SIP terminal process detects no ISUP MIME. A check is performed to verify that encapsulated ISUP is not expected, processing continues and passes notification to the ISUP terminal process. A default ISUP ANM message (without any parameters) is created and forwarded to the ST PH in the SS7 GSM. Finally, the message is sent to the originating switch.
25. The call path through the switch is established and remains available until the call is terminated.

Call Setup at TPS - No ISUP

This section takes a closer look at the processing and messaging performed by the TPS when acting as a gateway between an IP network and the PSTN.

Figure 3-22 Call Setup at TPS - Steps 1-6



1. An INVITE message arrives at, and is accepted by the SIP PH in the SIP GSM. This message includes:
   - IP address of OPS signaling point
   - IP address of TPS signaling point

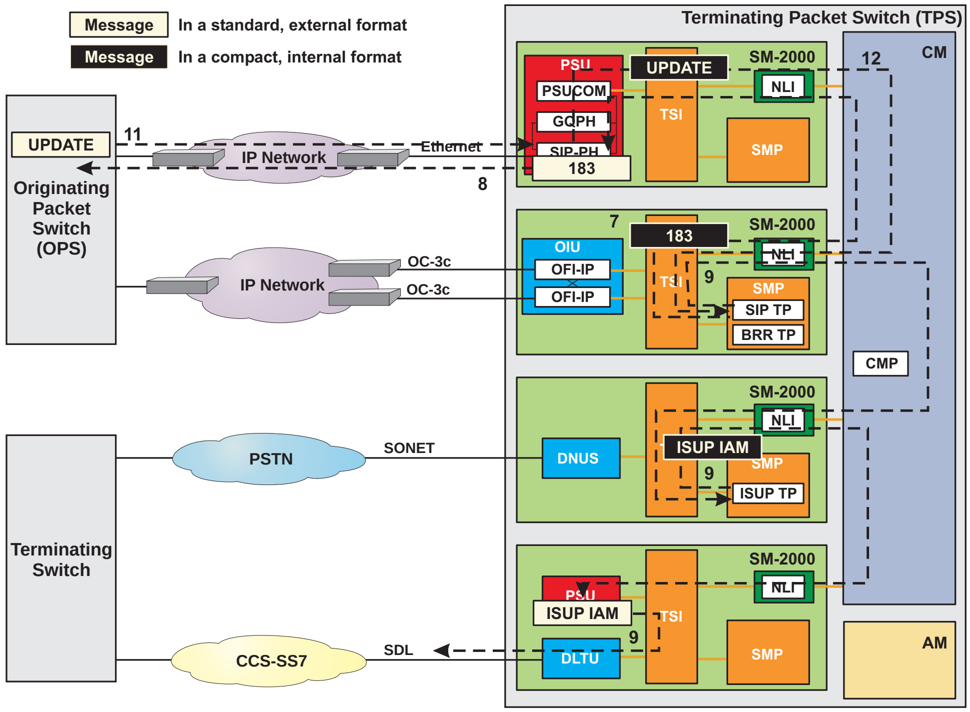
- IP address and port number for the OPS bearer resource
- A description of the requested session.

2. The SIP PH strips the SCTP and IP headers and compacts the message to the internal SIP format. The SIP PH then selects a switching module for the SIP terminal process and forwards the message.
3. The SIP system process receives the INVITE. It detects no ISUP MIME and checks to determine whether SFID 769 (SIP without Encapsulated ISUP) is **on**. If the SFID is active, as in this example, the SIP system process sets a flag to indicate subsequent ISUP messages should not be encapsulated, processing continues and the INVITE is sent to the SIP terminal process.
4. The SIP terminal process sees that ISUP encapsulation is not allowed and will default the mandatory parameters of the IAM. It also performs digit analysis to determine whether the call should be routed to a line or a trunk. In this example, it is determined that the call should be routed to an ISUP trunk. Additionally, the SIP terminal process determines that the call from the OPS must arrive at a bearer resource so it requests that the CMP select a bearer resource for the call.
5. The CMP:
   - determines that the call arrives at a SIP packet group,
   - selects a switching module for the bearer terminal process.

   The determined information as well as the IP address and port number of the OPS bearer resource are then forwarded to the selected IP bearer switching module.

6. The bearer terminal process selects an OFI and a port for the call. The bearer terminal process requests that the OFI open the selected port for this transaction. Finally, the IP address and port number of the selected resource are forwarded to the SIP terminal process.

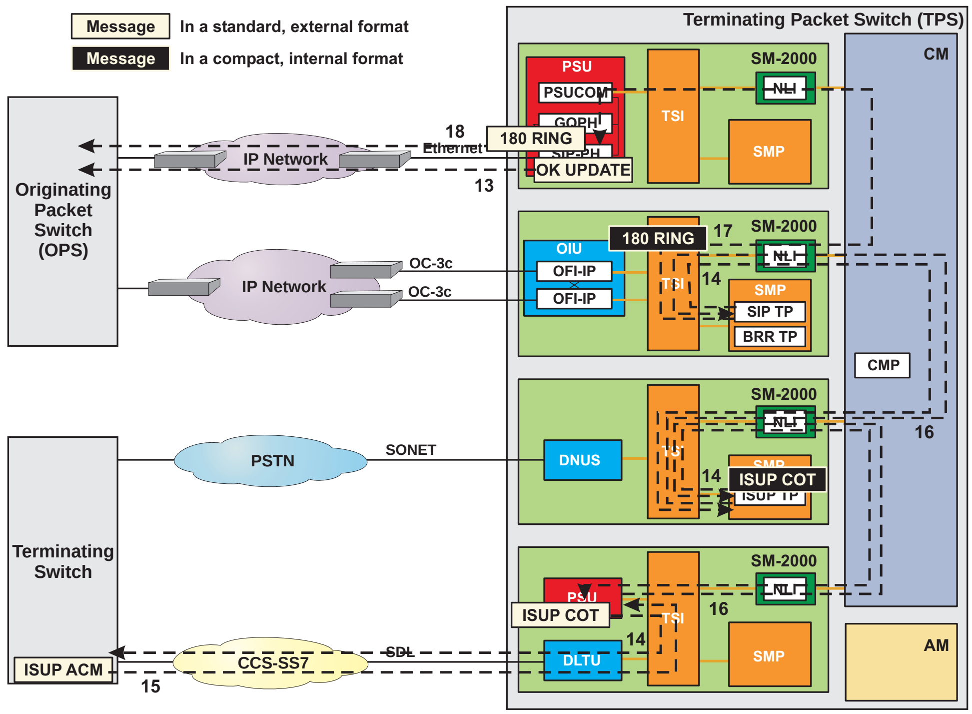
Figure 3-23 Call Setup at TPS - Steps 7-12



7. The SIP terminal process creates a 183 SESSION PROGRESS message. This message is in a compact, internal SIP format. This message contains the IP address and port for the terminating bearer resource. The SIP message is forwarded to the SIP PH in the SIP GSM.
8. The SIP PH expands the 183 SESSION PROGRESS message to standard external SIP format, adds the appropriate SCTP and IP headers, and forwards it to OPS via the IP signaling network.
9. The SIP terminal process passes notification to the ISUP terminal process. An ISUP IAM message is created and forwarded to the ST PH in the SS7 GSM. Finally, the message is sent to the terminating switch.
10. The TPS waits for the OPS to respond.

11. The SIP PH receives and accepts an UPDATE message from the OPS. The message informs the TPS that the OPS has opened the upstream port and that the bearer path is now available.
12. The SIP PH strips the SCTP and IP headers, compacts the message to the internal SIP format, and forwards the message to the SIP terminal process.

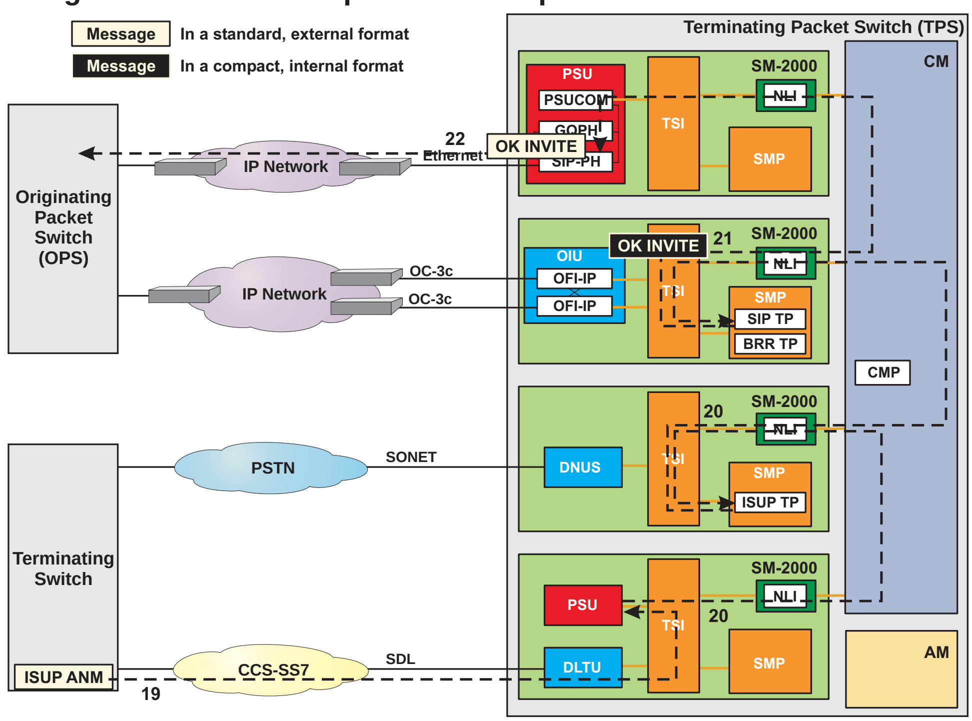
Figure 3-24 Call Setup at TPS - Steps 13-18



13. The SIP PH builds the 200 OK (UPDATE) message in standard, external SIP format, adds the appropriate SCTP and IP headers, and forwards it to OPS via the IP signaling network.
14. The SIP terminal process passes the updated call state to the ISUP terminal process. An ISUP COT message is formulated and forwarded to the ST PH in the SS7 GSM. Finally, the message is sent to the terminating switch.
15. An ISUP ACM message arrives at the TPS on a signaling data link (SDL). The message is sent from the trunk peripheral terminating the SDL to the ST PH in the SS7 GSM via nailed up timeslots.

16. The ST PH examines the message and forwards it to the ISUP terminal process. The message is then forwarded to the SIP terminal process.
17. The SIP terminal process develops a 180 RINGING message. A check is made to determine whether encapsulated ISUP is required. In this example, encapsulated ISUP is not required, therefore the terminal process discards the ISUP ACM MIME. This message is in a compact, internal SIP format. The SIP message is forwarded to the SIP PH in the SIP GSM.
18. The SIP PH receives and accepts the message with no ISUP MIME and expands the 180 RINGING message to standard, external SIP format, adds the appropriate SCTP and IP headers, and forwards it to OPS via the IP signaling network.

Figure 3-25 Call Setup at TPS - Steps 19-23



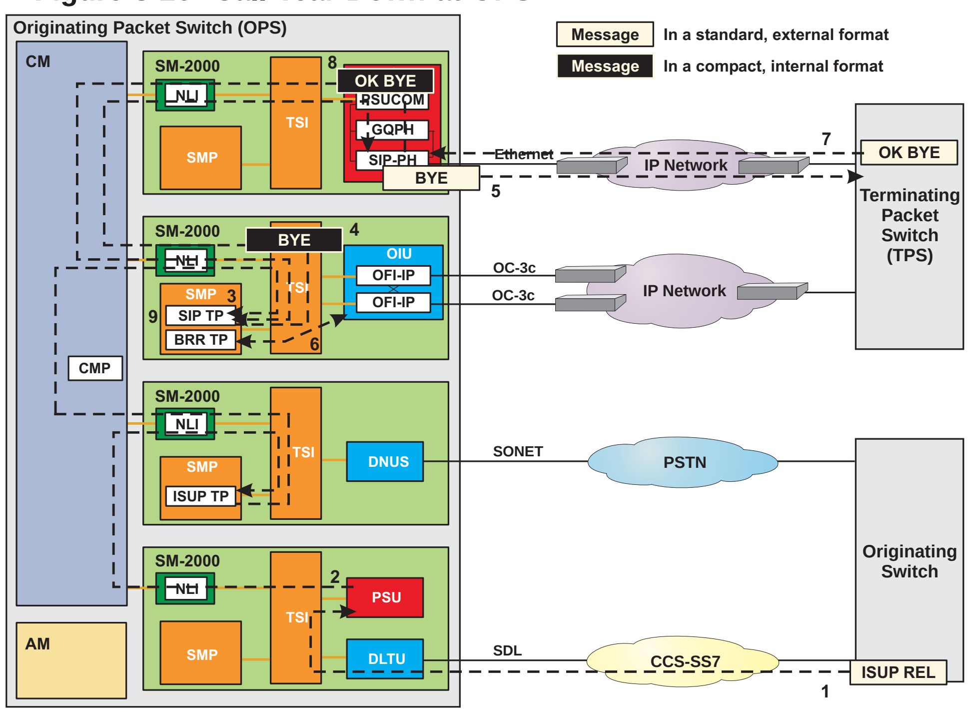
19. An ISUP ANM message arrives at the TPS on a signaling data link (SDL). The message is sent from the trunk peripheral terminating the SDL to the ST PH in the SS7 GSM via nailed up timeslots.

20. The ST PH examines the message and forwards it to the ISUP terminal process. The message is then forwarded to the SIP terminal process.
21. The SIP terminal process develops a 200 OK (INVITE) message. A check is made to determine whether encapsulated ISUP is required. In this example, encapsulated ISUP is not required, therefore the terminal process discards the ISUP ANM MIME. This message is in a compact, internal SIP format. The SIP message is forwarded to the SIP PH in the SIP GSM.
22. The SIP PH receives and accepts the message with no ISUP MIME and expands the 200 OK (INVITE) message to standard, external SIP format, adds the appropriate SCTP and IP headers, and forwards it to OPS via the IP signaling network.
23. The call path through the switch is established and remains available until the call is terminated.

Call Tear Down at OPS - No ISUP

This section takes a closer look at the processing and messaging completed by the OPS when acting as a gateway between the PSTN and an IP network.

Figure 3-26 Call Tear Down at OPS

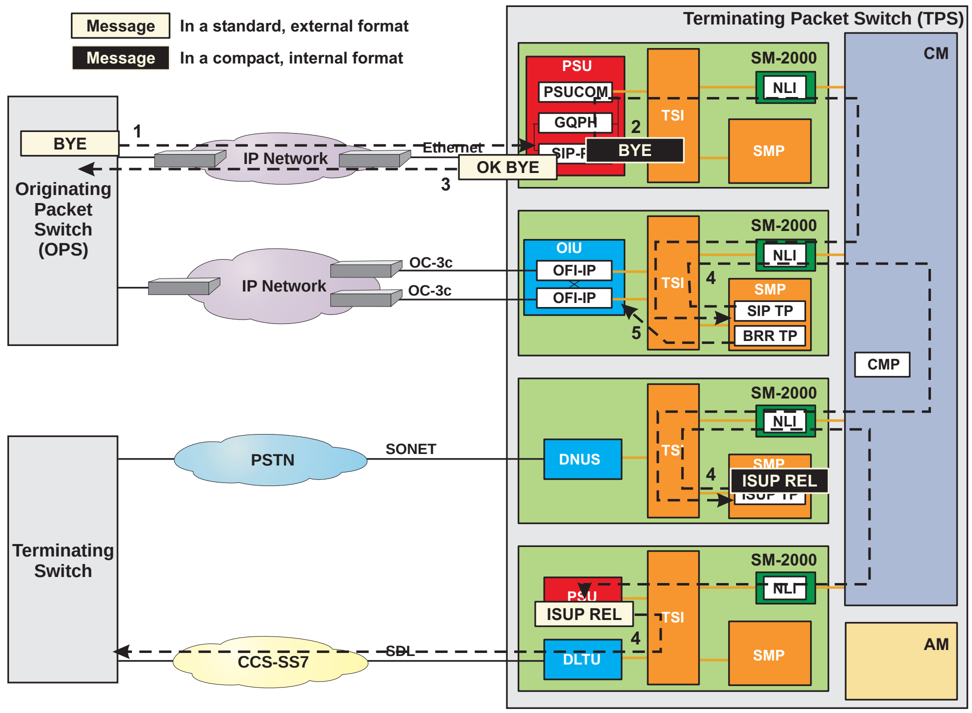


1. An ISUP REL message arrives at the OPS on a signaling data link (SDL). The message is sent from the trunk peripheral terminating the SDL to the ST PH in the SS7 GSM via nailed up timeslots.
2. The ST PH examines the message and forwards it to the ISUP terminal process. The message is then forwarded to the SIP terminal process.
3. Peripheral control in the IP bearer switching module detects discontinuity and informs the SIP terminal process of the change of state.
4. The SIP terminal process formulates a BYE message. At the end of processing, a check is made to determine whether encapsulated ISUP is allowed. If not, as in this example, the ISUP REL MIME is discarded. Processing continues and the message is forwarded to the SIP PH in the SIP GSM. This message is in a compact, internal SIP format. This message informs the TPS to tear down the call.
5. The SIP PH expands the BYE message to standard, external SIP format, without an ISUP MIME or SDP MIME, adds the appropriate SCTP and IP headers, and forwards it to the TPS via the IP signaling network.
6. The bearer terminal process takes steps to tear down the call by closing the port on the OFI and releasing used timeslots.
7. The SIP PH receives a 200 OK (BYE) message from the TPS. The message informs the OPS that it received the BYE message.
8. The SIP PH strips the SCTP and IP headers, compacts the message to the internal SIP format, and forwards the message to the SIP terminal process.
9. The SIP terminal process terminates any remaining processes dedicated to the call.
10. Call flow is complete.

Call Tear Down at TPS - No ISUP

This section takes a closer look at the processing and messaging completed by the TPS when acting as a gateway between the PSTN and an IP network.

Figure 3-27 Call Tear Down at TPS



1. The SIP PH receives and accepts the BYE message from the OPS. The message informs the TPS the calling party went on-hook.
2. The SIP PH strips the SCTP and IP headers, compacts the message to the internal SIP format, and forwards the message to the SIP terminal process. The SIP terminal process verifies that encapsulated ISUP is not required. If not required, as in this example, processing continues and a default ISUP REL is created.
3. The SIP PH builds the 200 OK (BYE) message in standard, external SIP format, adds the appropriate SCTP and IP headers, and forwards it to OPS via the IP signaling network.
4. The SIP terminal process passes the updated call state to the ISUP terminal process. An ISUP REL message is forwarded to the ST PH in the SS7 GSM. Finally, the message is sent to the terminating switch.

5. The SIP terminal process passes the updated call state to the bearer terminal process. The bearer terminal process takes steps to tear down the call by closing the port on the OFI and releasing used timeslots.
6. Call flow is complete.

□

SIP without Encapsulated ISUP - SIP Message Examples

This section includes example SIP messages to illustrate their format and content. Ethernet, IP, and SCTP headers were removed to focus on the SIP message and any SDP messages.

Figure 3-28 SIP INVITE {SDP}

```
INVITE
Session Initiation Protocol
    Request line: INVITE sip:+17952425001@10.11.2.25;transport=sctp;user=phone
SIP/2.0
        Method: INVITE
    Message Header
        t: <sip:+17952425001@10.11.2.25;transport=sctp;user=phone>
        f: <sip:+18152205000@10.11.2.9;transport=sctp;user=phone>;tag=0-02-014-
00014-1441
        i: 110675ba-ee530082-e05d75ce@LABM1
        CSeq: 1 INVITE
        v: SIP/2.0/SCTP 10.11.2.9:1;branch=z9hG4bK0140116.ebbe.00000001
        m: <10.11.2.9:1>
        Max-Forwards: 20
        Accept: application/SDP, multipart/mixed
        MIME-Version: 1.0
        c: application/SDP
        Allow: INVITE, ACK, BYE, CANCEL, UPDATE, INFO, OPTIONS
        Require: precondition
        l: 255
Session Description Protocol
    Session Description Protocol Version (v): 0
    Owner/Creator, Session Id (o): - 0 0 IN IP4 192.168.2.5
        Owner Username: -
        Session ID: 0
        Session Version: 0
        Owner Network Type: IN
        Owner Address Type: IP4
        Owner Address: 192.168.2.5
    Session Name (s): 0
    Connection Information (c): IN IP4 192.168.2.5
        Connection Network Type: IN
        Connection Address Type: IP4
        Connection Address: 192.168.2.5
    Time Description, active time (t): 0 0
        Session Start Time: 0
        Session Start Time: 0
    Media Description, name and address (m): audio 50614 RTP/AVP 0
        Media Type: audio
        Media Port: 50614
        Media Proto: RTP/AVP
        Media Format: 0
    Bandwidth Information (b): AS:64
        Bandwidth Modifier: AS
        Bandwidth Value: 64
```

```
    Media Attribute (a): ptime:30
        Media Attribute Fieldname: ptime
        Media Attribute Value: 30
    Media Attribute (a): rtpmap:0 PCMU/8000
        Media Attribute Fieldname: rtpmap
        Media Attribute Value: 0 PCMU/8000
    Media Attribute (a): curr:qos local none
        Media Attribute Fieldname: curr
        Media Attribute Value: qos local none
    Media Attribute (a): curr:qos remote none
        Media Attribute Fieldname: curr
        Media Attribute Value: qos remote none
    Media Attribute (a): des:qos mandatory local sendrecv
        Media Attribute Fieldname: des
        Media Attribute Value: qos mandatory local sendrecv
    Media Attribute (a): des:qos mandatory remote sendrecv
        Media Attribute Fieldname: des
        Media Attribute Value: qos mandatory remote sendrecv
```

Figure 3-29 SIP 183 SESSION PROGRESS {SDP}

```
183 Session Progress
Session Initiation Protocol
    Status line: SIP/2.0 183 Session Progress
        Status-Code: 183
    Message Header
        t: <sip:+17952425001@10.11.2.25;transport=sctp;user=phone>;tag=0-04-014-
40014-1443
        f: <sip:+18152205000@10.11.2.9;transport=sctp;user=phone>;tag=0-02-014-
00014-1441
        i: 110675ba-ee530082-e05d75ce@LABM1
        CSeq: 1 INVITE
        v: SIP/2.0/SCTP 10.11.2.9:1;branch=z9hG4bK0140116.ebbe.00000001
        MIME-Version: 1.0
        c: application/SDP
        Allow: INVITE, ACK, BYE, CANCEL, UPDATE, INFO, OPTIONS
        Require: precondition
        l: 285
Session Description Protocol
    Session Description Protocol Version (v): 0
    Owner/Creator, Session Id (o): - 0 0 IN IP4 192.168.2.13
        Owner Username: -
        Session ID: 0
        Session Version: 0
        Owner Network Type: IN
        Owner Address Type: IP4
        Owner Address: 192.168.2.13
    Session Name (s): 0
    Connection Information (c): IN IP4 192.168.2.13
        Connection Network Type: IN
        Connection Address Type: IP4
        Connection Address: 192.168.2.13
    Time Description, active time (t): 0 0
        Session Start Time: 0
        Session Start Time: 0
```

```
    Media Description, name and address (m): audio 52394 RTP/AVP 0
        Media Type: audio
        Media Port: 52394
        Media Proto: RTP/AVP
        Media Format: 0
    Bandwidth Information (b): AS:64
        Bandwidth Modifier: AS
        Bandwidth Value: 64
    Media Attribute (a): ptime:20
        Media Attribute Fieldname: ptime
        Media Attribute Value: 20
    Media Attribute (a): rtpmap:0 PCMU/8000
        Media Attribute Fieldname: rtpmap
        Media Attribute Value: 0 PCMU/8000
    Media Attribute (a): curr:qos local none
        Media Attribute Fieldname: curr
        Media Attribute Value: qos local none
    Media Attribute (a): curr:qos remote none
        Media Attribute Fieldname: curr
        Media Attribute Value: qos remote none
    Media Attribute (a): des:qos mandatory local sendrecv
        Media Attribute Fieldname: des
        Media Attribute Value: qos mandatory local sendrecv
    Media Attribute (a): des:qos mandatory remote sendrecv
        Media Attribute Fieldname: des
        Media Attribute Value: qos mandatory remote sendrecv
    Media Attribute (a): conf:qos remote sendrecv
        Media Attribute Fieldname: conf
        Media Attribute Value: qos remote sendrecv
```

Figure 3-30 SIP UPDATE {SDP}

```
UPDATE
Session Initiation Protocol
    Request line: UPDATE sip:+17952425001@10.11.2.25;transport=sctp;user=phone 
SIP/2.0
        Method: UPDATE
    Message Header
        t: <sip:+17952425001@10.11.2.25;transport=sctp;user=phone>;tag=0-04-014-
40014-1443
        f: <sip:+18152205000@10.11.2.9;transport=sctp;user=phone>;tag=0-02-014-
00014-1441
        i: 110675ba-ee530082-e05d75ce@LABM1
        CSeq: 2 UPDATE
        v: SIP/2.0/SCTP 10.11.2.9:1;branch=z9hG4bK0140116.ebbe.00000002
        m: <10.11.2.9:1>
        Max-Forwards: 20
        MIME-Version: 1.0
        c: application/SDP
        l: 259
Session Description Protocol
    Session Description Protocol Version (v): 0
    Owner/Creator, Session Id (o): - 0 0 IN IP4 192.168.2.5
        Owner Username: -
        Session ID: 0
```

```
            Session Version: 0
            Owner Network Type: IN
            Owner Address Type: IP4
            Owner Address: 192.168.2.5
        Session Name (s): 0
        Connection Information (c): IN IP4 192.168.2.5
            Connection Network Type: IN
            Connection Address Type: IP4
            Connection Address: 192.168.2.5
        Time Description, active time (t): 0 0
            Session Start Time: 0
            Session Start Time: 0
        Media Description, name and address (m): audio 50614 RTP/AVP 0
            Media Type: audio
            Media Port: 50614
            Media Proto: RTP/AVP
            Media Format: 0
        Bandwidth Information (b): AS:64
            Bandwidth Modifier: AS
            Bandwidth Value: 64
        Media Attribute (a): ptime:30
            Media Attribute Fieldname: ptime
            Media Attribute Value: 30
        Media Attribute (a): rtpmap:0 PCMU/8000
            Media Attribute Fieldname: rtpmap
            Media Attribute Value: 0 PCMU/8000
        Media Attribute (a): curr:qos local sendrecv
            Media Attribute Fieldname: curr
            Media Attribute Value: qos local sendrecv
        Media Attribute (a): curr:qos remote none
            Media Attribute Fieldname: curr
            Media Attribute Value: qos remote none
        Media Attribute (a): des:qos mandatory local sendrecv
            Media Attribute Fieldname: des
            Media Attribute Value: qos mandatory local sendrecv
        Media Attribute (a): des:qos mandatory remote sendrecv
            Media Attribute Fieldname: des
            Media Attribute Value: qos mandatory remote sendrecv
```

Figure 3-31 SIP 200 OK (UPDATE) {SDP}

```
    200 OK (UPDATE)
    Session Initiation Protocol
        Status line: SIP/2.0 200 OK
            Status-Code: 200
        Message Header
            t: <sip:+17952425001@10.11.2.25;transport=sctp;user=phone>;tag=0-04-014-
    40014-1443
            f: <sip:+18152205000@10.11.2.9;transport=sctp;user=phone>;tag=0-02-014-
    00014-1441
            i: 110675ba-ee530082-e05d75ce@LABM1
            CSeq: 2 UPDATE
            v: SIP/2.0/SCTP 10.11.2.9:1;branch=z9hG4bK0140116.ebbe.00000002
            m: <10.11.2.25:1>
            MIME-Version: 1.0
```

```
        c: application/SDP
        l: 265
Session Description Protocol
    Session Description Protocol Version (v): 0
    Owner/Creator, Session Id (o): - 0 0 IN IP4 192.168.2.13
        Owner Username: -
        Session ID: 0
        Session Version: 0
        Owner Network Type: IN
        Owner Address Type: IP4
        Owner Address: 192.168.2.13
    Session Name (s): 0
    Connection Information (c): IN IP4 192.168.2.13
        Connection Network Type: IN
        Connection Address Type: IP4
        Connection Address: 192.168.2.13
    Time Description, active time (t): 0 0
        Session Start Time: 0
        Session Start Time: 0
    Media Description, name and address (m): audio 52394 RTP/AVP 0
        Media Type: audio
        Media Port: 52394
        Media Proto: RTP/AVP
        Media Format: 0
    Bandwidth Information (b): AS:64
        Bandwidth Modifier: AS
        Bandwidth Value: 64
    Media Attribute (a): ptime:20
        Media Attribute Fieldname: ptime
        Media Attribute Value: 20
    Media Attribute (a): rtpmap:0 PCMU/8000
        Media Attribute Fieldname: rtpmap
        Media Attribute Value: 0 PCMU/8000
    Media Attribute (a): curr:qos local sendrecv
        Media Attribute Fieldname: curr
        Media Attribute Value: qos local sendrecv
    Media Attribute (a): curr:qos remote sendrecv
        Media Attribute Fieldname: curr
        Media Attribute Value: qos remote sendrecv
    Media Attribute (a): des:qos mandatory local sendrecv
        Media Attribute Fieldname: des
        Media Attribute Value: qos mandatory local sendrecv
    Media Attribute (a): des:qos mandatory remote sendrecv
        Media Attribute Fieldname: des
        Media Attribute Value: qos mandatory remote sendrecv
```

Figure 3-32 SIP 180 RINGING

```
180 RINGING
Session Initiation Protocol
    Status line: SIP/2.0 180 Ringing
        Status-Code: 180
    Message Header
        t: <sip:+17952425001@10.11.2.25;transport=sctp;user=phone>;tag=0-04-014-
40014-1443
```

```
        f: <sip:+18152205000@10.11.2.9;transport=sctp;user=phone>;tag=0-02-014-
00014-1441
        i: 110675ba-ee530082-e05d75ce@LABM1
        CSeq: 1 INVITE
        v: SIP/2.0/SCTP 10.11.2.9:1;branch=z9hG4bK0140116.ebbe.00000001
        Allow: INVITE, ACK, BYE, CANCEL, UPDATE, INFO, OPTIONS
        Require: precondition
        l: 0
```

Figure 3-33 200 OK (INVITE)

```
200 OK (INVITE)
Session Initiation Protocol
    Status line: SIP/2.0 200 OK
        Status-Code: 200
    Message Header
        t: <sip:+17952425001@10.11.2.25;transport=sctp;user=phone>;tag=0-04-014-
40014-1443
        f: <sip:+18152205000@10.11.2.9;transport=sctp;user=phone>;tag=0-02-014-
00014-1441
        i: 110675ba-ee530082-e05d75ce@LABM1
        CSeq: 1 INVITE
        v: SIP/2.0/SCTP 10.11.2.9:1;branch=z9hG4bK0140116.ebbe.00000001
        m: <10.11.2.25:1>
        Allow: INVITE, ACK, BYE, CANCEL, UPDATE, INFO, OPTIONS
        l: 0
```

Figure 3-34 SIP ACK

```
ACK
Session Initiation Protocol
    Request line: ACK sip:+17952425001@10.11.2.25;transport=sctp;user=phone SIP/
2.0
        Method: ACK
    Message Header
        t: <sip:+17952425001@10.11.2.25;transport=sctp;user=phone>;tag=0-04-014-
40014-1443
        f: <sip:+18152205000@10.11.2.9;transport=sctp;user=phone>;tag=0-02-014-
00014-1441
        i: 110675ba-ee530082-e05d75ce@LABM1
        CSeq: 1 ACK
        v: SIP/2.0/SCTP 10.11.2.9:1;branch=z9hG4bK0140116.ebbe.00000001
        Max-Forwards: 20
        l: 0
```

Figure 3-35 SIP BYE

```
BYE
Session Initiation Protocol
    Request line: BYE sip:+17952425001@10.11.2.25;transport=sctp;user=phone SIP/
2.0
        Method: BYE
```

```
    Message Header
        t: <sip:+17952425001@10.11.2.25;transport=sctp;user=phone>;tag=0-04-014-
40014-1443
        f: <sip:+18152205000@10.11.2.9;transport=sctp;user=phone>;tag=0-02-014-
00014-1441
        i: 110675ba-ee530082-e05d75ce@LABM1
        CSeq: 3 BYE
        v: SIP/2.0/SCTP 10.11.2.9:1;branch=z9hG4bK0140116.ebbe.00000003
        Max-Forwards: 20
        l: 0
```

Figure 3-36 200 OK (BYE)

```
200 OK (BYE)
Session Initiation Protocol
    Status line: SIP/2.0 200 OK
        Status-Code: 200
    Message Header
        t: <sip:+17952425001@10.11.2.25;transport=sctp;user=phone>;tag=0-04-014-
40014-1443
        f: <sip:+18152205000@10.11.2.9;transport=sctp;user=phone>;tag=0-02-014-
00014-1441
        i: 110675ba-ee530082-e05d75ce@LABM1
        CSeq: 3 BYE
        v: SIP/2.0/SCTP 10.11.2.9:1;branch=z9hG4bK0140116.ebbe.00000003
        l: 0
```

□

4 Engineering Considerations

Overview

Purpose

The purpose of this chapter is to specify the engineering rules and recommendations for the Session Initiation Protocol (SIP) for Packet Trunking feature.

As required, the following topics will be covered in each section:

- capacity,
- configuration requirements,
- constraints,
- recommendations, and
- equipage.

The final sections in this chapter will address the following:

- measurement reports,
- network management reports, and
- operational support system (OSS) impacts.

□

Switch Considerations

Overview This section discusses switch-level considerations that must be observed when engineering the SIP for Packet Trunking feature.

The following call models are used to determine the capacity requirements.

Configuration Model	***Traffic Distribution***
IP OIU-Packet SIP Tandem (IOST)	• ISUP trunk calls -> ISUP trunk calls (33%) • ISUP trunk calls -> SIP calls (33.5%) • SIP calls -> ISUP trunk calls (33.5%)
IP OIU-Packet SIP EQPOTS (IOSE)	• analog line calls -> analog line calls (20%) • analog line calls -> ISUP trunk calls (13%) • analog line calls -> SIP calls (27%) • ISUP trunk calls -> analog line calls (13%) • SIP calls -> analog line calls (27%)
PSTN Gateway Local/Toll (PSTNGW-LT)	• analog line calls -> analog line calls (20%) • analog line calls -> SIP trunk calls (25%) • SIP trunk calls -> analog line calls (25%) • ISUP trunk calls -> IP end point calls (5%) • IP end point calls -> ISUP trunk calls (5%) • SIP trunk calls -> ISUP trunk calls (10%) • ISUP trunk calls -> SIP trunk calls (10%)

Capacity

The *5ESS*® switch supports 2 million busy hour calls (BHC) and up to 128K simultaneous SIP calls, subject to available switch resources on the IOST, IOSE, and PSTNGW-LT models.

An office supports up to four SIP global switch modules (GSMs).

Configuration requirements

The SIP for Packet Trunking feature requires that an office be equipped with a quad link packet switch (QLPS). A *5E-XC*™ that supports the SIP for Packet Trunking feature must be equipped with a QLPS, SM-2000 switching modules, OIU-IP, and a PSU2 with DF2 and PHE2 and PH33 protocol handler packs. For information regarding the PSU2 requirements, refer to "Packet Switch Unit Model 2 (PSU2) Considerations" (4-6).

Note: This QLPS and PH33 are not required for DRM/VCDX.

Constraints

The SIP for Packet Trunking feature is not supported on the following platforms or configurations:

- Extended Switch Module (EXM),
- Compact Digital Exchange (CDX),
- Non SM-2000,
- Remote Switch Module (RSM), and
- Optically Integrated Remote Switching Module (ORM)

☐

Global Switching Module (GSM) Considerations

Overview

This section discusses the engineering considerations for the SIP global switching module (GSM).

Capacity

The GSM supports up to 48 SIP protocol handlers (PHs); however, the current Ethernet grounding plate design limits the maximum number of SIP PHs to 16.

The GSM supports up to a sum total of 16 Signaling System 7 (SS7) QPHs and General QLPS Protocol Handlers (GQPHs). An office supports up to a sum total of 59 SS7 QLPS protocol handlers (QPHs) and GQPHs.

Note:The DRM/VCDX does not support any QPHs, neither CCS nor GQPHs.

Table 4-1, “IOST model BHC capacities” (4-4) , Table 4-2, “IOSE model BHC capacities” (4-4) , and Table 4-3, “PSTNGW-LT model BHC capacities” (4-4) show BHC capacities for other interfacing hardware components.

Table 4-1 IOST model BHC capacities

Component	BHC
SMP60MM	120K (at 90% capacity)
SMP60MM	100K (at 75% capacity)
CORE700	480K (at 90% capacity)
CORE700	400K (at 75% capacity)

Table 4-2 IOSE model BHC capacities

Component	BHC
SMP60MM	100K (at 90% capacity)
SMP60MM	82K (at 75% capacity)
CORE700	300K (at 90% capacity)
CORE700	268K (at 75% capacity)

Table 4-3 PSTNGW-LT model BHC capacities

Component	BHC
SMP60MM	100K (at 90% capacity)

Table 4-3 PSTNGW-LT model BHC capacities (continued)

Component	BHC
SMP60MM	83K (at 75% capacity)
CORE700	300K (at 90% capacity)
CORE700	249K (at 75% capacity)

Configuration requirements

The SIP for Packet Trunking feature requires that the SIP GSM:

- be an SM-2000, and
- contain a packet switch unit model 2 (PSU2).
- DRM IP trunking requires that the SM-2000 be a CORE700 SM, because of all the memory that is required when all the functionality (signaling, call processing, and bearer) is combined onto a single SM.

Earlier versions of the switching module and packet switch unit ***do not*** support SIP for Packet Trunking.

Recommendations

Though the GSM can be used as a bearer switching module (SM), it is recommended that the GSM not be engineered as the bearer SM for reliability, recovery, and performance reasons.

Note: This does not apply to the DRM/VCDX. Since there is only one SM, is necessary to support all of the signaling and bearer functionality on the same SM.

□

Packet Switch Unit Model 2 (PSU2) Considerations

Overview

This section discusses the engineering considerations for the packet switch unit model 2 (PSU2) to support SIP for Packet Trunking.

Configuration requirements

The SIP for Packet Trunking feature:

- is supported only on the first PSU2 of a double PSU2 configuration (only applies to wireless applications),
- requires PHE2s and PH33s, and
- requires a data fanout 2 (DF2) on each shelf equipped with PHE2s and/or PH33s.

Note: The PH33s are not required on the DRM/VCDX.

Earlier versions of PHs, for example PH22, PH3, etc., do not support SIP for Packet Trunking.

Recommendations

Although there are no restrictions on mixing PH types, it is recommended not to mix applications on the same GSM.

For increased reliability, any loadshared PHs such as GQPHs should be equipped across:

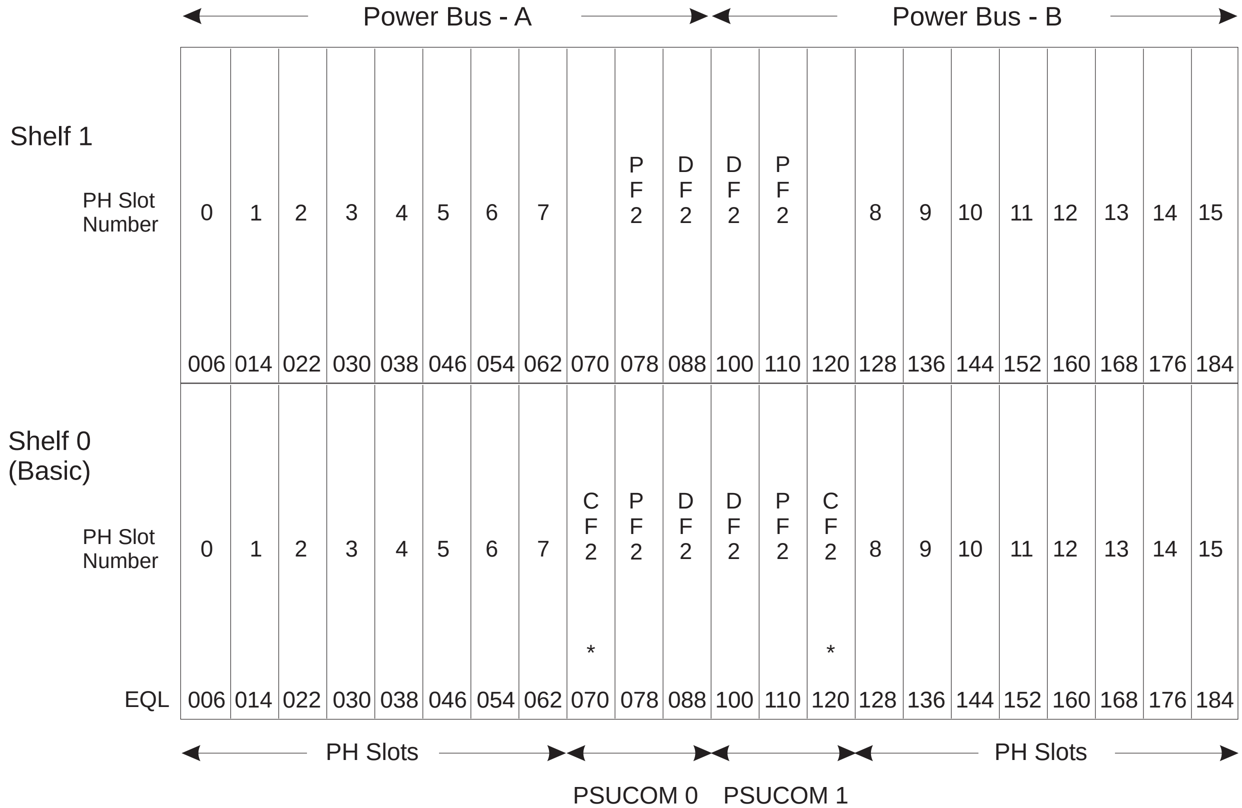
- separate fuse groups,
- power buses, and
- PSU2 shelves

Note: These recommendations do not apply to DRM/VCDX - there is only one SM, and there are not any loadshared GQPHs.

when possible. Refer to the *Administration and Engineering Guidelines*, 235-070-100 document for additional PH engineering considerations.

Figure 4-1, “Two shelf PSU2” (4-7) layout can be referenced when reviewing Table 4-4, “PH equipage - one shelf” (4-7) and Table 4-5, “PH equipage - two shelves” (4-8).

Figure 4-1 Two shelf PSU2



* Note: the CF2 packs are located in the basic shelf only.

Refer to Table 4-4, "PH equipage - one shelf" (4-7) for a suggested equipage. This equipage spreads the PH packs across power buses and fuse groups. It also provides efficient circuit pack cooling.

Note: This table provides equipage for the first eight PHs, when additional PHs are equipped in the shelf, the pattern is expanded to include the remaining slots.

Table 4-4 PH equipage - one shelf

Number of PHs	Horizontal EQL
1	062
2	062, 128
3	062, 128, 030
4	062, 128, 030, 160
5	062, 128, 030, 160, 046

Table 4-4 PH equipage - one shelf (continued)

Number of PHs	Horizontal EQL
6	062, 128, 030, 160, 046, 144
7	062, 128, 030, 160, 046, 144, 014
8	062, 128, 030, 160, 046, 144, 014, 176

When equipping more than one shelf, the pattern is slightly different. Table 4-5, “PH equipage - two shelves” (4-8) provides a partial pattern to follow when equipping two shelves. When additional PHs are added, the same pattern is expanded as required.

Table 4-5 PH equipage - two shelves

PHs per Shelf	First shelf Horizontal EQL	Second shelf Horizontal EQL
1	062	128
2	062, 128	128, 062
3	062, 128, 030	128, 062, 160
4	062, 128, 030, 160	128, 062, 160, 030
5	062, 128, 030, 160, 046	128, 062, 160, 030, 144
6	062, 128, 030, 160, 046, 144	128, 062, 160, 030, 144, 046
7	062, 128, 030, 160, 046, 144, 014	128, 062, 160, 030, 144, 046, 176
8	062, 128, 030, 160, 046, 144, 014, 176	128, 062, 160, 030, 144, 046, 176, 014

When additional shelves are added, the above pattern is extended as required.

□

Session Initiation Protocol - Protocol Handler (SIP PH) Considerations

Overview

This section discusses the engineering considerations for the SIP protocol handler (PH).

Capacity

When SIP signaling uses precondition procedures and is supported by the SCTP transport layer, as in the base SIP for Packet Trunking feature, the SIP PH can process approximately 350K calls per hour, with a small number of associations terminated on the SIP PH. When SIP signaling operates without precondition procedures and with the UDP transport layer, the capacity is considerably reduced.

The SIP PH supports one full duplex 100BaseT Ethernet link. The link can support approximately 2.65 million calls per hour depending on the average message size, however, the overall engineered capacity is primarily limited by the processor capacity of the SIP PH.

Configuration requirements

Table 4-6, "SIP PH Configuration" (4-9) illustrates the engineering configurations required to support a SIP PH.

Table 4-6 SIP PH Configuration

SM	Global SM-2000
Terminations	single full duplex PHE2 Ethernet Link
PSU Type	PSU2 equipped with DF2
PH Hardware	TN13 with an LLE2 100BaseT Ethernet Paddle board

SIP PHs can only be provisioned in PSU 0 if a dual-PSU configuration exists on a SIP GSM. There are no constraints on coexistence with other types of protocol handlers aside from the number of physical circuit pack slots available in the PSU2 shelf.

Recommendations

SIP PHs in the same processor group should be assigned to different power buses and fuse groups, if possible. Refer to PSU2 "Recommendations" (4-6) for additional detail.

It is recommended, for reliability purposes, to engineer the SIP PH processor groups as duplex (i.e., each processor group has a pair of SIP PHs). As a duplex processor group, one SIP PH acts as the serving SIP PH and the other acts as the non-serving SIP PH.

Equipage Table 4-7, "SIP PH Equipage" (4-10) provides the SIP PH hardware required for IP trunking based on the office BHC load.

Table 4-7 SIP PH Equipage

Required Hardware	Office BHC Load			
	0 - 350K	***350K - 700K***	***700K - 1.05M***	***1.05M - 1.4M***
SIP PH Pairs	1	2	3	4
PHE2 (TN13) Packs	2	4	6	8
LLE2 Paddle boards	2	4	6	8

General QLPS Protocol Handler (GQPH) Considerations

Overview

This section discusses the engineering considerations for the general QLPS protocol handler (PH).

Note: The GQPH section does not apply to DRM/VCDX.

Capacity

The GQPH processes up to 2.3 million calls per hour for both the IOST and IOSE models. It processes up to 2 million calls per hour for the PSTNGW-LT model.

Configuration requirements

Table 4-8, "GQPH Configuration" (4-11) illustrates the engineering configurations required to support a GQPH.

Table 4-8 GQPH Configuration

SM	Global SM-2000
DF2/NCT2 Timeslots	24 per QLPS side X 2 QLPS sides
PSU Type	PSU2 equipped with DF2 packs
PH Hardware	TN113

Recommendations

PH33 boards used for GQPHs should be spread across different power buses and fuse groups for added reliability. Refer to PSU2 "Recommendations" (4-6) for additional detail.

It is recommended that GQPHs are engineered active load shared. Each shelf that has one or more GQPH channel groups assigned should have at least one spare PH33 to allow softswitching of the GQPH channel group to the spare PH33 board during routine exercise (REX) and application of software updates requiring PH file replacement.

Equipage

The number of GQPHs is dependent on the number of SS7 QPHs that may be sharing the same GSM PSU2. The sum total of SS7 QPHs and GQPHs may not exceed 16 per SM-2000 or 59 per office.

Each GQPH supports two 24-Network Control Timing Link 2 (NCT2) Timeslot Qpipes, one to each QLPS side. This is twice the amount (12 timeslots) that an SS7 QPH supports. The maximum engineerable number of GQPHs and QPHs is subject to the 256 timeslots available on the primary NLI to each QLPS side. However, of those 256 timeslots, 16 timeslots are reserved, leaving 240 available timeslots. Refer to Table 4-9, "Control - messaging timeslots" (4-12) for additional information.

Table 4-9 Control - messaging timeslots

Primary NLI timeslots available for control/messaging	256
Reserved timeslots	16
Timeslots required per GQPH	24
Timeslots required per QPH	12

Table 4-10, "GQPH Equipage" (4-12) provides the GQPH hardware required to support the maximum message load for IP trunking, based on the office configuration. One TN113 is recommended as a spare for each GQPH equipped shelf.

Table 4-10 GQPH Equipage

Required Hardware	Configuration	
	IOST	***IOSE***
PH33 (TN113) Packs	2	2

Packet Group Considerations

Overview

This section discusses packet group engineering considerations.

Equipage

One (or more) SIP packet group(s) must be provisioned from the *5E-XC*™ to each switch in the network to which it must communicate directly for SIP packet trunking. Each SIP packet group must be supported uniquely either by one association set when the SCTP transport layer is used, or by one UDP path when the UDP transport layer is used. The transport layer used (SCTP verses UDP) places different limitations on the number of SIP packet groups that can be provisioned.

□

Association Set Considerations

Overview This section discusses association set engineering considerations.

Equipage The number of association sets is determined by the number of SIP packet groups that use SCTP as the transport layer to the far office. There is a one-to-one correspondence between SIP packet groups and association sets when SCTP transport is used.

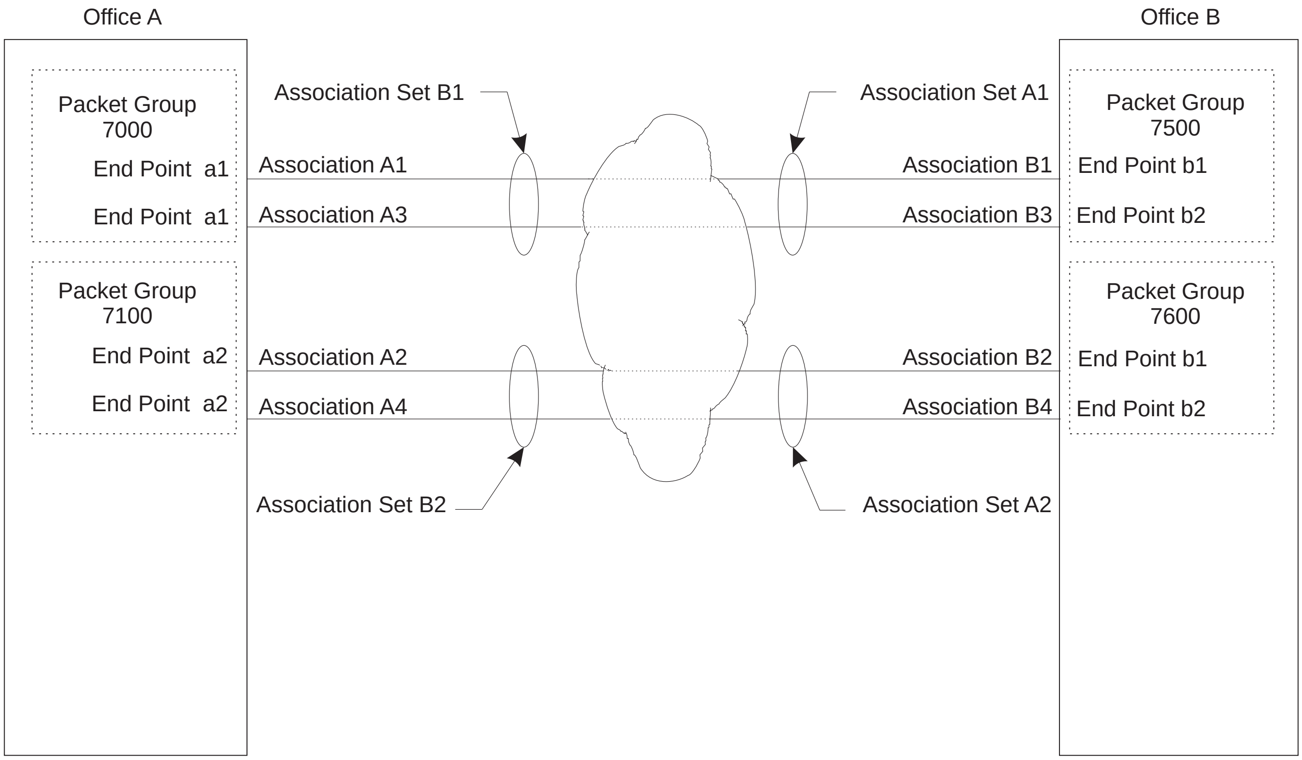
An association set must contain at least one SCTP association, and may contain up to 64 associations. The number of associations in an association set is determined by the number of unique SCTP endpoints in the switches at each end of the packet group. If there is one SCTP endpoint in each switch, then there can only be one association in one association set for one packet group between the two switches. If either (or both ends) support more than one SCTP endpoint, additional associations may be provisioned between each pair of near and far SCTP endpoints to allow for loadsharing and additional reliability when the associations are grouped in an association set. There is an office-wide maximum limit of 1023 associations. The maximum number of associations could all be provisioned on a single SCTP near endpoint if there were the maximum number of different far-end SCTP endpoints with which to communicate; however, each association consumes resources on the SIP PH.

When engineering association sets between two offices, the offices should agree on their interface. Specifically:

- define the association sets that are needed between the offices,
- agree which packet group uses which association set, and
- have each office independently provision packet groups to use the agreed upon association sets.

See the example in Figure 4-2, "Engineering Association Sets" (4-15).

Figure 4-2 Engineering Association Sets



In Figure 4-2, "Engineering Association Sets" (4-15), Association Set B1, in office A, consists of two associations:

- association A1, and
- association A3

The above information is defined in RC/V.

□

Stream Control Transmission Protocol (SCTP) End Point Considerations

Overview This section discusses the engineering considerations for stream control transmission protocol (SCTP) end points.

Equipage The number of SCTP endpoints per SIP PH processor group is one. Therefore, engineering the number of SCTP endpoints is dependent on the number of SIP PHs.

□

User Datagram Protocol (UDP) Path Considerations

Overview

This section discusses the engineering considerations for user datagram protocol (UDP) paths.

Equipage

The number of UDP paths is determined by the number of SIP packet groups that use UDP as the transport layer to the far office. There is a one-to-one correspondence between UDP-supported SIP packet groups and UDP paths. There is an office-wide maximum limit of 128 UDP paths. The maximum number of UDP paths could all be provisioned on a single SIP processor group if there were the maximum number of different far-end IP addresses with which to communicate.

□

IP Addressing Schemes

Overview

Diverse IP routing allows for signaling paths to take diverse routes over the IP network so that a single point of failure within the IP network does not block signaling traffic between two offices. For a more reliable transport, it is recommended that IP network diverse IP routing be engineered.

Note: Diverse IP routing only applies when multihomed SCTP endpoints are used to transport SIP messages. UDP does not support multihoming.

□

Multihoming

Overview

Multihoming has the added advantage of providing multiple paths in the IP network thus increasing the reliability. In the *5E-XC*™ implementation, supporting multihoming requires two IP addresses per endpoint. In order to take advantage of the multihoming capability, the customer's IP network must be engineered in such a way that packets routed using the two IP addresses traverse different physical paths in the network.

Note: Diverse IP routing only applies when multi-homed SCTP endpoints are used to transport SIP messages. UDP does not support multi-homing.

□

Measurements

Dedicated SIP reports

The SIP for Packet Trunking feature provides three sections to the TRFC30 report. These sections include:

- **Section 209: SIP Measurements**- provides measurements for the SIP signaling traffic report.
- **Section 210: SCTP Measurements**- provides measurements for the SCTP protocol traffic report.
- **Section 211: PKTGRP Measurements**- provides measurements for the Packet Group traffic report.

Refer to the *Output Messages*, 235-600-750, document for additional information on these traffic reports.

SIP counters in other reports

In addition to the dedicated SIP sections, the following existing TRFC30 counters include counters that accounted for SIP signaling traffic:

- counters in Section 253 (Call MIX) related to IP traffic type
- counter in Section 3 (POVFL) when no packet resources for voice channels exist
- the following counters of section DSLG for GQPH message traffic
  - OVLD
  - OCCUP
  - MSGIN
  - MSGOUT
- the following counters of Section PSUCHAN for GQPH message traffic:
  - XMTOCC
  - RCVOCC
  - XMT95
  - RCV95
  - MSGSXMT
  - MSGRCV

- the following counters of Section DSLG for SIP PH message traffic:
  - OVLD
  - OCCUP

Refer to the *Output Messages*, 235-600-750, document for additional information on these traffic reports.

□

Network Management

Overview

Network management (NM) for SIP for Packet Trunking is similar to SS7 signaling. However, Trunk Reservation (TR) and Service Selection Trunk Reservation (SSTR) are not supported for outgoing packet transported calls.

SIP for Packet Trunking adds four new sections to the NM 5-minute package report. The new packages include:

- the PKTGRP package that provides the counts on SIP call traffic,
- the SIPT package that provides the counts on SIP signaling traffic,
- the SCTP package that provides the counts on SCTP layer traffic measured on a per office basis, and
- the internet control message protocol (ICMP) package that provides the counts on the low layer protocols of SIP signaling.

Refer to the *Output Messages*, 235-600-750, document for additional information on these traffic reports.

□

Operational Support Systems

Overview

The operational support systems (OSS) impacted by SIP for Packet Trunking are:

- IP address management
- SCTP association management
- Input commands for the following entities:
  - Processor Group
  - Ethernet link
  - SCTP endpoints
  - SCTP associations
  - GQPH links and QPipes
- State information for the following entities:
  - PSU2 PH
  - Ethernet link
  - SCTP near endpoint
  - SCTP associations
  - GQPH Links and QPipes
- Signaling abnormalities
- Inter-office testing for SCTP associations
- Traffic measurement counts
- Performance monitoring data
- Hardware inventory data for optical facility interface (OFI) circuit pack

□

5 Provisioning

Overview

Purpose

This chapter contains the procedures for provisioning the Session Initiation Protocol (SIP) application in the *5ESS*® switch.

All provisioning procedures described in this section are performed in the RC/V menu mode. The RC/V menu mode can be entered from the RC/V terminal or a Supplemental Trunk Line Workstation (STLWS).

The *5ESS*® switch Recent Change and Verify (RC/V) interface provides access to the database through views of the Office Dependent Data (ODD). Only those views and steps necessary to provision SIP are covered in this chapter. For complete descriptions of RC/V views or more information about recent change and verify, refer to *Recent Change Reference*, 235-118-258 and *Recent Change Procedures*, 235-118-251.

The Provisioning section does not include the procedures for provisioning traffic measurements.

At the end of this chapter there are additional Provisioning procedures that may be required for growing and/or changing your already existing SIP signaling network. Please reference the "Add/Change SIP Elements within an Existing SIP Network" (5-93) procedures.

□

Provisioning Sequences

Introduction

This section contains the startup procedures for provisioning SM connectivity and SIP signaling. Figure 5-1, "SM Connectivity Provisioning Flowchart" (5-3), Figure 5-2, "Signaling Provisioning Flowchart for SIP with SCTP Transport" (5-5), and Figure 5-3, "Signaling Provisioning Flowchart for SIP with UDP Transport" (5-7) contain the steps that can be executed synchronously and asynchronously. Each section contains example scenarios that can be used for provisioning the SIP application.

Note: The SM connectivity provisioning does not apply to DRM/VCDX.

In the flowcharts, the arrows indicate which RC/V views must be provisioned before the next RC/V can be provisioned. For example, in Figure 5-2, "Signaling Provisioning Flowchart for SIP with SCTP Transport" (5-5) the boxes with arrows pointing to the box with Association (RC/V 33.22) must be executed before the Association (RC/V 33.22) can be executed. Therefore the SCTP Near Endpoint (RC/V 33.19), and Far Endpoint (RC/V 33.21) views must be executed before the Association (RC/V 33.22) view.

The following RC/V views are listed in each flowchart but are executed only once. These views are listed in each flowchart to show they must be provisioned before other views.

- Secured Feature (RC/V 8.22)
- SIP GSM (RC/V 5.80)
- Equipment Subunit Protocol Handler (RC/V 22.4)
- PSU Protocol Handler Channel Group Assignment (RC/V 22.16)

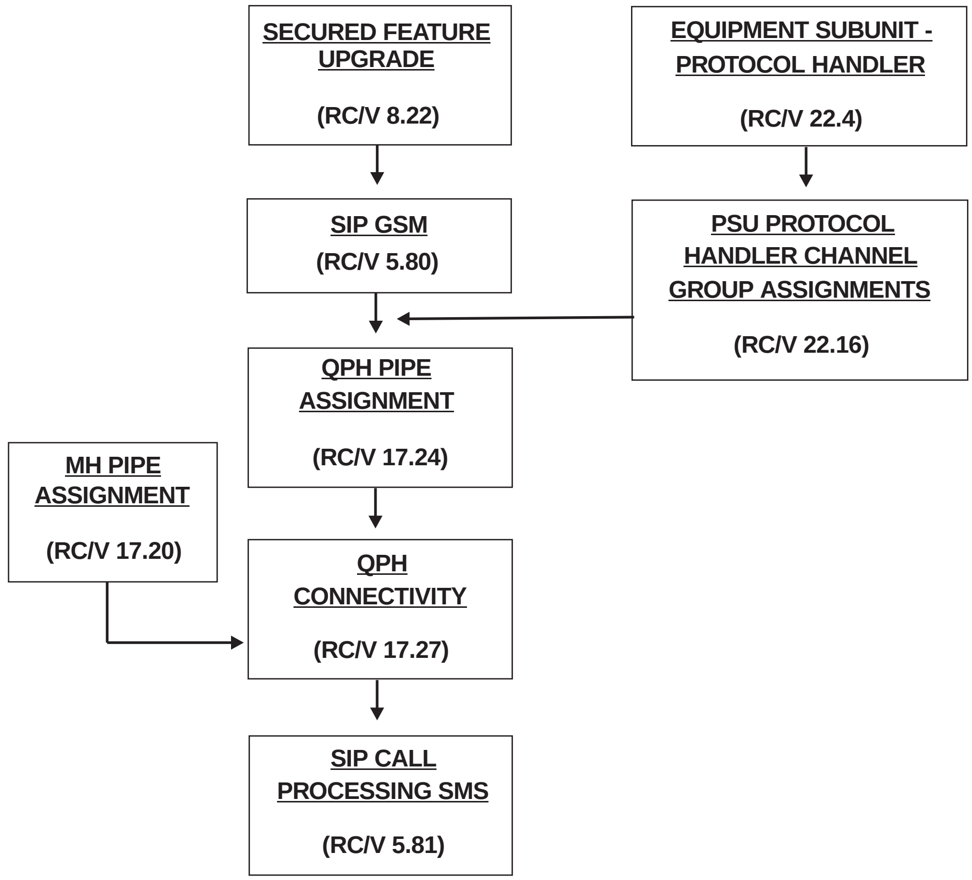
Some provisioning may already be complete prior to provisioning the SIP application. For example the MH Pipe Assignment RC/V 17.20 is executed only once when an SM-2000 is added to the network. This procedure does not need to be executed if another SM-2000 is not being added.

Provisioning SM Connectivity

Figure 5-1, "SM Connectivity Provisioning Flowchart" (5-3) displays the provisioning sequence for SM Connectivity.

Note: The SM connectivity provisioning does not apply to DRM/VCDX.

Figure 5-1 SM Connectivity Provisioning Flowchart



The flowchart indicates that some steps can be executed synchronously and asynchronously. Below is an acceptable sequence of the steps displayed in Figure 5-1, "SM Connectivity Provisioning Flowchart" (5-3) .

1. "Feature Activation (RC/V 8.22)" (5-13)
2. "Insert SIP Global SM (RC/V 5.80)" (5-16)
3. "Define Protocol Handler (RC/V 22.4)" (5-18)
4. "PH Channel Group Assignment (RC/V 22.16)" (5-19)
5. " Assign QPH Pipe (RC/V 17.24)" (5-23)
6. "Assign MH Pipe (RC/V 17.20)" (5-26)
7. "Update GSM - Non-GSM Communication (RC/V 17.27)" (5-28)
8. "Insert Call Processing SM (RC/V 5.81)" (5-31)

Other sequences could be utilized as long as they comply with the flow illustrated in Figure 5-1, "SM Connectivity Provisioning Flowchart" (5-3).

Provisioning SIP with SCTP Transport

Figure 5-2 Signaling Provisioning Flowchart for SIP with SCTP Transport

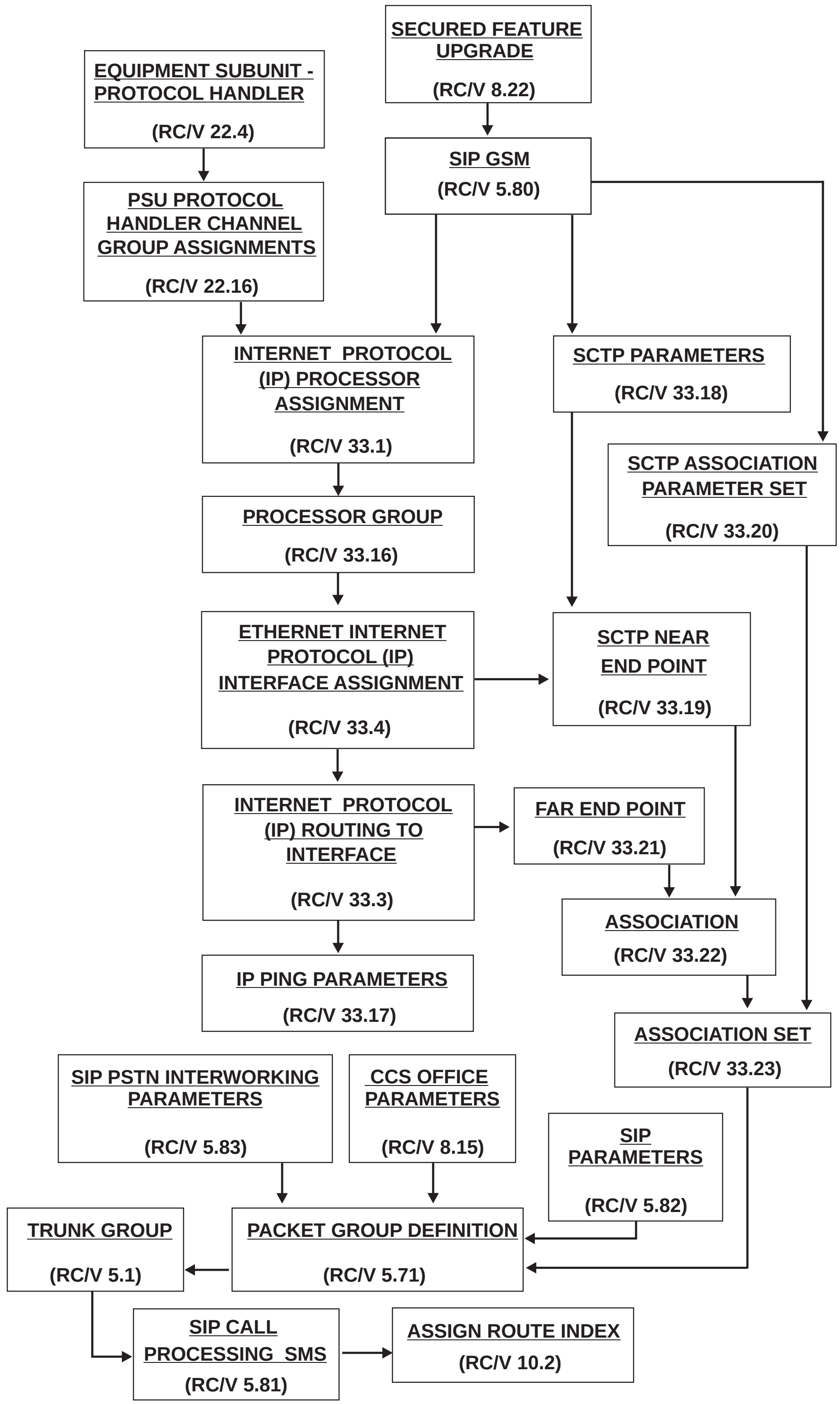


Figure 5-2, "Signaling Provisioning Flowchart for SIP with SCTP Transport" (5-5) displays the provisioning sequence for SIP with SCTP Transport.

The flowchart indicates that some steps can be executed synchronously and asynchronously. Below is an acceptable sequence of the steps displayed in Figure 5-2, "Signaling Provisioning Flowchart for SIP with SCTP Transport" (5-5) .

1. "Insert SIP Global SM (RC/V 5.80)" (5-16)
2. "Define Protocol Handler (RC/V 22.4)" (5-18)
3. "PH Channel Group Assignment (RC/V 22.16)" (5-19)
4. "Assign PH Channel Group as IP Processor (RC/V 33.1)" (5-34)
5. "Insert Processor Group (RC/V 33.16)" (5-38)
6. "Insert Ethernet - IP Interface (RC/V 33.4)" (5-41)
7. "Insert IP Routing (RC/V 33.3)" (5-44)
8. "Insert Router Pinging (RC/V 33.17)" (5-47)
9. "Insert SCTP Endpoint Timers and Protocol Parameters (RC/V 33.18)" (5-53)
10. "Insert SCTP Near Endpoints (RC/V 33.19)" (5-56))
11. "Insert SCTP Far Endpoints (RC/V 33.21)" (5-59)
12. "Insert SCTP Association-related Protocol Parameters (RC/V 33.20)" (5-62)
13. " Insert SCTP Association (RC/V 33.22)" (5-65)
14. " Insert SCTP Association Set (RC/V 33.23)" (5-68)
15. "Update CCS Office Parameters (RC/V 8.15)" (5-71)
16. "Insert SIP PSTN Interworking Parameter Set (RC/V 5.83)" (5-74)
17. "Insert SIP Parameters (RC/V 5.82)" (5-79)
18. "Insert Packet Group (RC/V 5.71)" (5-82)
19. "Assign Trunk Group (RC/V 5.1)" (5-85)
20. "Enable INVITE Requests (RC/V 5.81)" (5-88)
21. "Assign Route Index (RC/V 10.2)" (5-91)

Other sequences could be utilized as long as they comply with the flow illustrated in Figure 5-2, "Signaling Provisioning Flowchart for SIP with SCTP Transport" (5-5).

Provisioning SIP with UDP Transport

Figure 5-3 Signaling Provisioning Flowchart for SIP with UDP Transport

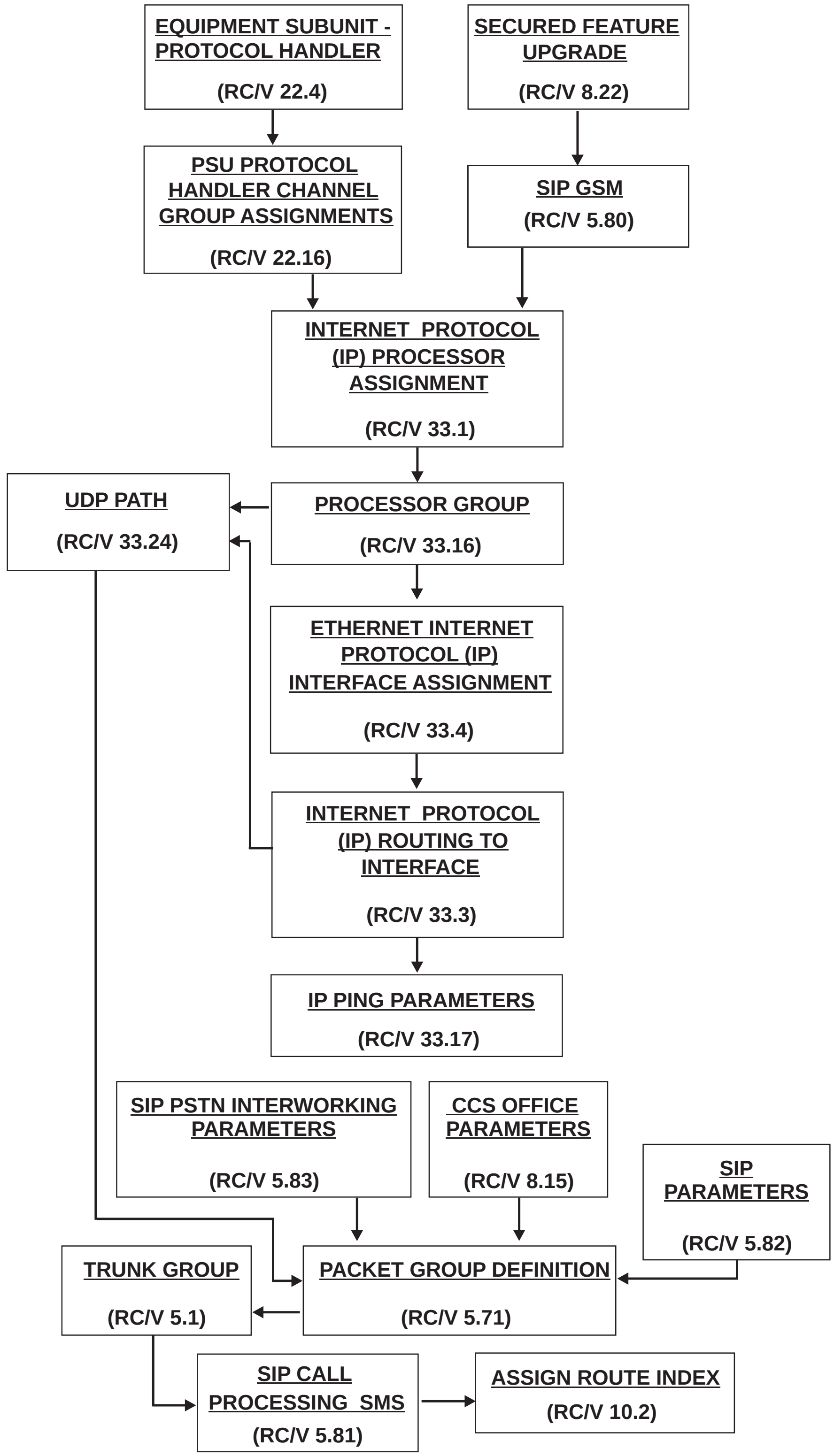


Figure 5-3, “Signaling Provisioning Flowchart for SIP with UDP Transport” (5-7) displays the provisioning sequence for SIP with UDP Transport.

The flowchart indicates that some steps can be executed synchronously and asynchronously. Below is an acceptable sequence of the steps displayed in Figure 5-3, “Signaling Provisioning Flowchart for SIP with UDP Transport” (5-7).

1. “Insert SIP Global SM (RC/V 5.80)” (5-16)
2. “Define Protocol Handler (RC/V 22.4)” (5-18)
3. “PH Channel Group Assignment (RC/V 22.16)” (5-19)
4. “Assign PH Channel Group as IP Processor (RC/V 33.1)” (5-34)
5. “Insert Processor Group (RC/V 33.16)” (5-38)
6. “Insert Ethernet - IP Interface (RC/V 33.4)” (5-41)
7. “Insert IP Routing (RC/V 33.3)” (5-44)
8. “Insert Router Pinging (RC/V 33.17)” (5-47)
9. “Insert UDP Path (RC/V 33.24)” (5-50)
10. “Update CCS Office Parameters (RC/V 8.15)” (5-71)
11. “Insert SIP PSTN Interworking Parameter Set (RC/V 5.83)” (5-74)
12. “Insert SIP Parameters (RC/V 5.82)” (5-79)
13. “Insert Packet Group (RC/V 5.71)” (5-82)
14. “Assign Trunk Group (RC/V 5.1)” (5-85)
15. “Enable INVITE Requests (RC/V 5.81)” (5-88)
16. “Assign Route Index (RC/V 10.2)” (5-91)

Other sequences could be utilized as long as they comply with the flow illustrated in Figure 5-3, “Signaling Provisioning Flowchart for SIP with UDP Transport” (5-7).

Prerequisites

This section contains a list of prerequisites for installing the SIP for Packet Trunking - NAR feature. All items should be verified prior to executing any step in the provisioning sequence.

- Examine office records to determine a suitable SM-2000 to serve as the SIP Global SM. The SM-2000 must be equipped with a PSU2. The absolute minimum equipage required is a PSU2 with one or more shelves with DF2 and at least five empty slots for installing PHs. The PHs must be available in PSU-0 if dual-PSU (two PSU2s) is equipped. SIP cannot be provisioned on PSU-1 of a dual PSU. If there are no DF2-equipped shelves with enough empty slots, it will be necessary to order a new PSU2 shelf with a DF2.
  Ideally, the SM-2000 selected should not be equipped with any trunks or lines or other call-processing roles. This reduces the chances of interference. The SM-2000 selected could be an existing SS7 GSM, but for additional reliability, it is recommended that the two global PSU2 functionalities are separate. If the SIP GSM goes through a full init or a duplex PSUCOM failure, calls can still be routed through SS7 when the two global PSU2 functionalities are separate.
- Examine office records to determine SM for which SM-2000s are provisioned with OIU-IP/OFI protection groups to serve as IP bearer endpoints.
- Examine the office records and traffic measurements occupancy reports to determine which SM-2000s have the available capacity to serve as SIP call processing SMs.
- Identify the adjacent layer 2 switches and/or IP routers external to the *5ESS*® switch to which the Ethernet links carrying the SIP signaling traffic will be connected for the *5ESS*® switch as well as for the adjacent routers.
- Identify the necessary IP addresses for the *5ESS*® switch as well as for the adjacent routers. There can be a maximum of two external IP addresses assigned to the Ethernet/IP interface for each processor group provisioned on the SIP GSM when SCTP transport is used. When UDP transport is used, only one IP address is used for the Ethernet/IP interface of a processor group.
- When SCTP transport is used, identify the IP addresses and SCTP ports on the other SIP-enabled switches in the network to which the *5ESS*® switch will be connecting SIP calls. A maximum of two far end addresses per association are supported.

- When UDP transport is used, identify the IP addresses and UDP ports on the other SIP-enabled switches in the network to which the *5ESS*® switch will be connecting SIP calls. UDP transport supports only one IP address at each end.
- Verify that sufficient PH hardware is available to install in the global PSU2 and for replacement boards. The absolute minimum is two TN13 packs with LLE2 100BaseT Ethernet paddleboards and three TN113 packs with no paddleboards.

Note:All the above recommendations for selecting SMs don’t apply to DRM/VCDX, since there is only the one SM.

Disclaimer IP management is the customer’s responsibility.

□

Select and Prepare Terminal

Purpose

This procedure contains instructions for selecting and preparing a terminal for RC/V activities.

When to use

This procedure is executed before any RC/V activities.

Related information

Refer to 235-118-251, Recent Change Procedures, for information on the RC/V system.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS), or
- Recent Change and Verify (RC/V) terminal

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 or later software release.

Required Information

- No information is required.

Required Conditions

- No conditions are required.

Procedure for STLWS terminal

1 Locate a STLWS. Verify the workstation is in command mode then type and enter at the CMD prompt:

`196`

Result:

RECENT CHANGE AND VERIFY view is displayed and cursor is located at PRINT OPTION.

2 Enable the print option.

`Y`

Result:

Cursor is located at DETAIL OPTION.

3 Disable the detail option.

`N`

Result:

Cursor is located at the VERBOSE OPTION.

4 Enable the verbose option.

`Y`

Result:

RECENT CHANGE AND VERIFY CLASSES menu page is displayed.

5 Stop. You have completed this procedure.

END OF STEPS

Procedure for RC/V terminal

1 Locate a RC/V type of terminal and enter the command:

`RCV:MENU:APPRC,VERBOSE,PRINT;`

Result:

RECENT CHANGE AND VERIFY CLASSES menu page is displayed.

2 Stop. You have completed this procedure.

END OF STEPS

□

Feature Activation (RC/V 8.22)

Purpose

This section describes the procedure for activating Secured Feature IDs (SFIDs) associated with SIP for Packet Trunking and related features.

When to use

The following SFIDs must be activated using RC/V 8.22 as described:

- SFID 684 is activated before any other provisioning procedures are attempted. This procedure is executed only once per office unless the feature has been intentionally deactivated.
- SFID 769 is activated if SIP without Encapsulated ISUP is supported.

Related information

Refer to 235-118-258, Recent Change Reference, for information on the RC/V system.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking - NAR feature software.
- If UDP transport is to be used for SIP signaling, and/or SIP is to be supported without precondition procedures, the *5E-XC*™ must be running a 5E16.2 FR5 software release with software for these enhancements to the SIP for Packet Trunking feature.
- If SIP without Encapsulated ISUP is supported, the *5E-XC*™ must be running on the 5E16.2 FR6 software release or later.
- If IP trunking is supported on a DRM, the *5ESS*® switch must be running on the 5E16.2 FR9 software release or later.

Required Information

- Secure Feature ID
- Module
- Password from Lucent Technologies Customer Account Team

Required Conditions

- The conditions in the prerequisite section must be completed.

Procedure

1 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

2 Type and enter the RC/V form number.

```
8.22
```

Result:

```
Enter Database Operation R=Review, U=Update:
```

3 Type and enter the update command.

```
U
```

Result:

SECURED FEATURE UPGRADE page is displayed and the cursor is located at FEATURE ID field

4 Using the RC/V 8.22 form as a guide, type and enter the indicated values for each field.

- *FEATURE ID - 684
- MODULE - OFC
- *PASSWD - 8-character password from Lucent Technologies.
- ACTIVE - Y

Result:

```
Enter Update, Change, Validate, screen#, or Print:
```

5 Enter the update command.

```
U
```

Result:

`Updating...form updated`

6 If SIP is to be supported:

- without Encapsulated ISUP, repeat steps 4 and 5 for SFID 769.

7 Type and enter the previous screen command.

`<`

Result:

8.0 OFFICE MISC. & ALARM VIEWS is displayed

8 If desired, type and enter the quit command to exit the RC/V system.

`Q`

Result:

The RC/V session is terminated.

9 Stop. You have completed this procedure.

END OF STEPS

Insert SIP Global SM (RC/V 5.80)

Purpose The SIP Global SM view (RC/V 5.80) is used to insert, review, update, and delete the SIP Global Switching module.

When to use This procedure is executed to identify an SM to be used as a SIP Global SM.

Related information Refer to 235-118-258, Recent Change Reference, for information on the RC/V system.

Admonishments None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release with the SIP for Packet Trunking - NAR feature software.

Required Information

- GSM number

Required Conditions

- Secured Feature 684 is enabled (RC/V 8.22)

Procedure

1 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

2 Type and enter the RC/V form number.

```
5.80
```

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

3 Type and enter the insert command.

`I`

Result:

The SIP GLOBAL SM page is displayed and the cursor is located at the GSM field

4 Using the RC/V 5.80 form as a guide, type and enter the parameters.

- *GSM- GSM number

5 Enter the insert command.

`I`

Result:

`Inserting...form inserted`

6 Type and enter the previous screen command.

<

Result:

The TRUNK VIEWS page is displayed.

7 If desired, type and enter the quit command to exit the RC/V system.

`Q`

Result:

The RC/V session is terminated.

8 Stop. You have completed this procedure.

END OF STEPS

Define Protocol Handler (RC/V 22.4)

Purpose

The Equipment Subunit Protocol Handler view (22.4) defines the hardware for the protocol handlers equipped on the shelf.

When to use

Use this procedure to define PH if a PH does not exist in the switch.

Related information

Refer to 235-118-258, Recent Change Reference, for information on the RC/V system.

Refer to 235-105-231, Hardware Change Procedures - Growth for information on PH growth.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release with the SIP for Packet Trunking - NAR feature software.
- The PHs must be installed in the shelf of the PSU2.

Required Information

- Switching Module (SM) Number
- Packet Switching Interface Unit Number
- Packet Switching Unit Shelf Number
- Position

Required Conditions

- The CRIT PSU (critical PSU2) must be set to N on RC/V 22.2.

Procedure

1 Refer to the growth procedures documentation.

Reference:

235-105-231, Hardware Change Procedures - Growth

END OF STEPS

□

PH Channel Group Assignment (RC/V 22.16)

Purpose

The PSU2 Protocol Handler Channel Group Assignments view (22.16) defines channel groups for protocol handlers equipped on a shelf. A channel group is a logical identifier for a protocol handler.

When to use

Use this procedure to define a new SIP PH and GQPH.

Related information

Refer to 235-118-258, Recent Change Reference, for information on the RC/V system.

Refer to 235-105-231, Hardware Change Procedures - Growth.

Admonishments

None.

Before you begin

Protocol Handlers must be grown in on the Equipment Subunit -- Protocol Handler view (22.4) and made operational before Channel Groups can be assigned on this view.

There is no direct link between the POSITION number on view 22.4 and the GRP number on this view, but there must be enough Protocol Handlers on view 22.4 to support all the Channel Groups and to account for spares.

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release with the SIP for Packet Trunking - NAR feature software.
- The PHs must be grown in the shelf of the PSU2.

Required Information

- Switching Module (SM) Number
- Packet Switch Unit Number
- Packet Switch Unit Shelf Number

Required Conditions

- Protocol Handler is defined (RC/V 22.4)

Procedure

1 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

2 Type and enter the RC/V form number.

`22.16`

Result:

`Enter Database Operation R=Review,U=Update:`

3 Type and enter the update command.

`U`

Result:

Screen 1 of the PSU PROTOCOL HANDLER CHANNEL GROUP ASSIGNMENTS page is displayed and the cursor is located at the SM field.

4 Using the RC/V 22.16 form as a guide, type and enter the indicated values for each field on screen 1.

- *SM - Switching Module (1-192)
- *PSU - Packet Switch Unit (0)
- *PSU SHELF - PSU2 shelf number (0-4)
- *VIRTUAL - Virtual PSU2 shelf (Y/N)

5 On screen 1 of the PSU PROTOCOL HANDLER CHANNEL GROUP ASSIGNMENTS page, determine how many PHE2 or PH33 are available.

- The number next to AVAILABLE PHE2 or AVAILABLE PH33 is the number of available PHE2s and PH33s, respectively.
- If more PHE2s or PH33s are needed refer to 235-105-231, Hardware Change Procedures - Growth.

6 Proceed to screen 2.

`2`

Result:

Screen 2 of the PSU PROTOCOL HANDLER CHANNEL GROUP ASSIGNMENTS page appears.

---

7 To add a PHE2 proceed to step 8.

To add a PHE33 proceed to step 9, not applicable for DRM/VCDX..

---

8 Using screen 2 and/or 3 of RC/V 22.16 as a guide, type and enter the following information for the PH.

- PH TYPE - PHE2
- AUTO ASSIGN - N
- RMK (optional) - i.e., enter SIPT

Result:

The PHE2 channel group is listed in the CHANNEL GROUPS (CHNLIST) list.

---

9 Using screen 2 and/or 3 of RC/V 22.16 as a guide type and enter the following information for the PH.

- PH TYPE - PH33
- AUTO ASSIGN - N
- RMK (optional) - i.e., enter GQPH

NOTE: This step should be skipped for DRM/VCDX..

Result:

PH33 has been provisioned.

---

10 Repeat steps 7-9 for each PH.

---

11 Enter the update command.

```
U
```

Result:

```
Updating...form updated
```

12 Type and enter the previous screen command.

<

Result:

ISDN -- EQUIPMENT VIEWS page is displayed.

13 If desired, type and enter the quit command to exit the RC/V system.

`Q`

Result:

The RC/V session is terminated.

14 Stop. You have completed this procedure.

END OF STEPS

Assign QPH Pipe (RC/V 17.24)

Purpose

The QPH Pipe Assignment (RC/V 17.24) defines the GQPH pipe connecting the QPH or GQPH to the QLPS network.

When to use

Use this procedure to provision the GQPH pipes for SIPT service.

Repeat individual steps of this procedure as required.

NOTE: This step is not needed for DRM/VCDX..

Related information

Refer to 235-118-258, Recent Change Reference, for information on the RC/V system and to 235-105-231, Hardware Change Procedures - Growth.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release with the SIP for Packet Trunking - NAR feature software.

Required Information

- GSM number
- PSU Shelf
- Channel Group
- QLPS Network
- Service Type

Required Conditions

- Growth procedures are complete
- Secured Feature 684 is enabled (RC/V 8.22)
- SIP GSM is provisioned (RC/V 5.80)
- Protocol Handler is defined (RC/V 22.4)
- Channel Groups are provisioned for PH33 (RC/V 22.16)

Procedure

1 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

2 Type and enter the RC/V form number.

```
17.24
```

Result:

```
Enter Database Operation I=Insert, R=Review,
   D=Delete:
```

3 Type and enter the insert command.

```
I
```

Result:

The QPH PIPE ASSIGNMENT page is displayed and the cursor is located at the GLOBAL SM field.

4 Using the RC/V 17.24 form as a guide, type and enter the parameters.

- *GLOBAL SM
- *PSU SHELF
- *CHANNEL GROUP
- *QLPS NETWORK
- #SERVICE TYPE - SIPT

5 Enter the insert command.

```
I
```

Result:

```
Inserting...form inserted
```

6 Type and enter the previous screen command.

```
<
```

Result:

CM MODULES VIEW page is displayed.

7 If desired, type and enter the quit command to exit the RC/V system.

```
Q
```

Result:

The RC/V session is terminated.

8 Proceed to MCC page 118X,Y,X to restore GQPH pipe.

Request the status of the GQPH QPipe:

```
OP:STATUS,GQPHPIPE,QPIPE=a-b-c-d-e;
```

Where:

a = Global switch module number

b = Packet switch unit 2 number

c = PSU2 shelf number

d = GQPH channel group number

e = QLPS network number

Examine the output.

Restore the GQPHPIPE to the ACTIVE state

```
RST:GQPHQPIPE=a-b-c-d-e
```

Where:

a = Global switch module number

b = Packet switch unit 2 number

c = PSU2 shelf number

d = GQPH channel group number

e = QLPS network number

For additional information, refer to the SIP Maintenance Considerations procedure, *Resolve GQPH QPipe Problems,* 235-200-118.

9 Stop. You have completed this procedure.

END OF STEPS

Assign MH Pipe (RC/V 17.20)

Purpose

The MH Pipe view (RC/V 17.20) is used to define a Message Handler (MH) pipe assignment for Quad-Link Packet Switch (QLPS). The MH pipe is the path from a QLPS to a MH.

When to use

Use this procedure to define a MH pipe when a SM-2000 exists.

NOTE: This step is not needed for DRM/VCDX..

Related information

Refer to 235-118-258, Recent Change Reference, for information on the RC/V system and to 235-105-231, Hardware Change Procedures - Growth.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- None

Required Information

- Switching Module (SM)

Required Conditions

- None.

Procedure

1 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

2 Type and enter the RC/V form number.

```
17.20
```

Result:

```
Enter Database Operation I=Insert,
```

```
R=Review, D=Delete:
```

3 Type and enter the insert command.

```
I
```

Result:

The MH PIPE ASSIGNMENT page is displayed.

4 Using the RC/V 17.20 form as a guide, type and enter the parameters.

- *QLPS NETWORK
- *SM2000 SM

5 Type and enter the insert command.

```
I
```

Result:

```
Inserting...form inserted
```

6 Type and enter the previous screen command.

```
<
```

Result:

The CM MODULE VIEWS page is displayed.

7 If desired, type and enter the quit command to exit the RC/V system.

```
Q
```

Result:

The RC/V session is terminated.

8 Stop. You have completed this procedure.

END OF STEPS

Update GSM - Non-GSM Communication (RC/V 17.27)

Purpose

The General GSM/NGSM Connectivity view (RC/V 17.27) defines the logical links between the GSM and Non-GSM.

When to use

Use this procedure to define connectivity between a GSM and non-GSM for intraswitch networking.

Only SM-2000s are allowed to connect to a GQPH.

Repeat individual steps of this procedure as required.

NOTE: This step is not needed for DRM/VCDX..

Related information

Refer to 235-118-258, Recent Change Reference, for information on the RC/V system and to 235-105-231, Hardware Change Procedures - Growth.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release with the SIP for Packet Trunking - NAR feature software.

Required Information

- Service Type.
- GSM number
- Non-GSM(s) number

Required Conditions

- Growth procedures are complete
- Secured Feature 684 is enabled (RC/V 8.22)
- SIP GSM is provisioned (RC/V 5.80)
- Protocol Handler is defined (RC/V 22.4)
- Channel groups are provisioned (RC/V 22.16)

- QPH pipes are provisioned (RC/V 17.24)
- MH pipes are provisioned (RC/V 17.20)

Procedure

1 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

2 Type and enter the RC/V form number.

`17.27`

Result:

`Enter Database Operation R=Review,U=Update:`

3 Type and enter the update command.

`U`

Result:

The GENERAL GSM/NGSM CONNECTIVITY page is displayed and the cursor is located at the GSM field.

4 Using the RC/V 17.27 form as a guide, type and enter the parameters.

- *GSM - GSM number
- *SERVICE TYPE - SIPT
- NON-GSM LIST - non-GSM numbers

5 Enter the update command.

`U`

Result:

`Updating...form updated`

6 Type and enter the previous screen command.

`<`

Result:

The CM MODULE VIEWS page is displayed.

7 If desired, type and enter the quit command to exit the RC/V system.

`Q`

Result:

The RC/V session is terminated.

8 Stop. You have completed this procedure.

END OF STEPS

Insert Call Processing SM (RC/V 5.81)

Purpose

The SIP-T Call Processing SMs view (RC/V 5.81) defines SM-2000s that can serve as a call processing SM for SIP calls.

When to use

Use this procedure to disable or enable an SM-2000 from processing SIP calls.

Repeat individual steps of this procedure as required.

Related information

Refer to 235-118-258, Recent Change Reference, for information on the RC/V system.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release with the SIP for Packet Trunking - NAR feature software.

Required Information

- SM-2000 module

Required Conditions

- Secured Feature 684 is enabled (RC/V 8.22)
- SIP GSM is provisioned (RC/V 5.80)
- Protocol Handler is defined (RC/V 22.4)
- Channel groups are provisioned (RC/V 22.16)
- QPH pipes are provisioned (RC/V 17.24)
- MH pipes are provisioned (RC/V 17.20)
- Global SM/GQPH connectivity is provisioned (RC/V 17.27)

NOTE: The last 3 steps are not required when provisioning a DRM/VCDX SM for SIP call processing.

Procedure

1 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

2 Type and enter the RC/V form number.

`5.81`

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

3 Type and enter the insert command.

`I`

Result:

The SIP-T CALL PROCESSING SMS page is displayed and the cursor is located at the GSM field.

4 Using the RC/V 5.81 form as a guide, type and enter the parameters.

- * GSM
- SM2K MODULE - number
- ROUTE - N

5 Enter the insert command.

`I`

Result:

`Inserting...form inserted`

NOTE: If GSM# equals SM2K#, a Warning message is displayed.

```
"SM2K Module in SIP-T SMS list should not equal GSM. This
     is not applicable for DRM/VCDX office"
```

If GSM should/does equal non-GSM enter "I" to ignore the warning and continue.

6 Type and enter the previous screen command.

<

Result:

The TRUNKS VIEWS page is displayed.

7 If desired, type and enter the quit command to exit the RC/V system.

`Q`

Result:

The RC/V session is terminated.

8 Stop. You have completed this procedure.

END OF STEPS

Assign PH Channel Group as IP Processor (RC/V 33.1)

Purpose

The Internet Protocol (IP) Processor Assignment view (RC/V 33.1) assigns IP addresses and parameters to a processor (PHE2).

When to use

Use this procedure to define IP parameters on a processor.

Repeat individual steps of this procedure as required.

Related information

Refer to 235-118-258, Recent Change Reference, for information on the RC/V system.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release with the SIP for Packet Trunking - NAR feature software.

Required Information

- Processor ID
- Processor type
- Shelf number
- Channel Group number
- Local IP address
- IP subnet mask

Required Conditions

- Protocol Handler is defined (RC/V 22.4)
- Channel Group is defined (RC/V 22.16)

Procedure

1 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

2 Type and enter the RC/V form number.

```
33.1
```

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

3 Type and enter the insert command.

```
I
```

Result:

Screen 1 of the INTERNET PROTOCOL (IP) PROCESSOR ASSIGNMENT page is displayed and the cursor is located at PROCESSOR ID field.

4 Using screen 1 of RC/V 33.1 form as a guide, type and enter the indicated values for each field.

- *PROCESSOR ID - SM number
- PROCESSOR TYPE - (For this SIP Platform document, PROCESSOR TYPE is PH.)
- *QUALIFIER 2 - For a PH, this field is the PSU community address (COM ADDR) found on RC/V 22.2, Packet Switch Unit).
- *QUALIFIER 3 - The 1 digit shelf number and 2 digit channel group number concatenated).
- LOCAL IP ADDR - customer defined
- IP SUBNET MASK - customer defined

NOTE: The LOCAL IP ADDR on this view is internal to the *5E-XC*™, and is not the IP address known externally to the SIP signaling network. The external IP address(es) are assigned to the Ethernet-IP interface on RC/V 33.4, in a separate procedure.

5 After entering all LOCAL IP ADDRs and IP SUBNET MASKs;

Type > and enter to access command line.

6 Type and enter screen 2.

`2`

Result:

Screen 2 of the INTERNET PROTOCOL (IP) PROCESSOR ASSIGNMENT page is displayed and the cursor is located at the REASSEM TIMER field.

7 Using screen 2 of RC/V 33.1 change the following field.

- #MTU ENABLE - Y

8 Proceed to screen 3.

`3`

Result:

Screen 3 of the INTERNET PROTOCOL (IP) PROCESSOR ASSIGNMENT page is displayed and the cursor is located at the ICMP ERR GEN field

9 Using screen 3 of RC/V 33.1 modify the following fields.

- #ICMP ERR GEN
- #IP FRAGMENT - Y
- #MTU INTVL AFT FAIL - 600

10 Enter the insert command.

`I`

Result:

`Warnings are displayed.`

11 Enter the ignore command for each Warning.

`I`

Result:

`Ignoring...form inserted.`

12 Type and enter the previous page command.

<

Result:

INTERNET PROTOCOL VIEWS page is displayed.

13 If desired, type and enter the quit command to exit the RC/V system.

`Q`

Result:

The RC/V session is terminated.

14 Stop. You have completed this procedure.

END OF STEPS

Insert Processor Group (RC/V 33.16)

Purpose

The Processor Group view (RC/V 33.16) defines a processor group. A processor group identifies one or two SIP PHs. If two SIP PHs operate as a pair with serving and non-serving roles, the serving and non-serving roles are chosen dynamically.

When to use

Use this procedure to define a new processor group.

Repeat individual steps of this procedure as required.

Related information

Refer to 235-118-258, Recent Change Reference, for information on the RC/V system.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release with the SIP for Packet Trunking - NAR feature software.

Required Information

- Processor Group Number - identifies a processor group
- SM - identifies SM where a processor group is located
- Application type - identifies the type of processor group
- PSU number - identifies the PSU2
- PM Group - identifies a performance monitoring group
- PH information
  - PSU shelf - identifies PSU2 shelf
  - Channel group - identifies PH’s Channel Group on the shelf
  - Position - identifies physical slot of PH on the shelf that is defined by a hardware procedure
  - PHE link - identifies a link number to a PHE2

Required Conditions

- Hardware Growth Procedures are complete.
- Protocol Handler slot is defined (RC/V 22.4)
- Channel Group is assigned a hardware type of PHE2 (RC/V 22.16)
- Channel Group is assigned as IP processor
- PM Group is defined (RC/V 20.32)

Procedure

1 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

2 Type and enter the RC/V form number.

```
33.16
```

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

3 Type and enter the insert command.

```
I
```

Result:

PROCESSOR GROUP page is displayed and the cursor is located at PCR GRP field

4 Using the RC/V 33.16 form as a guide, type and enter the indicated values for each field.

- *PCR GRP
- *SM - SIP GSM number
- APPL TYPE - SIPT
- PSU - 0
- PM GROUP - PM group name

5 Enter the PH Information for each PH under PROTOCOL HANDLERS.

- SHELF - shelf number
- CHGRP - channel group
- POSITION - physical slot of PH on shelf
- PHE LINK - link number to PHE2

6 Enter the insert command.

`I`

Result:

`Inserting...form inserted`

NOTE: View MCC page 118X,Y,X to see corresponding PH status change to Degraded (DGR).

7 Type and enter the previous screen command.

<

Result:

INTERNET PROTOCOL VIEWS page is displayed.

8 If desired, type and enter the quit command to exit the RC/V system.

`Q`

Result:

The RC/V session is terminated.

9 Stop. You have completed this procedure.

END OF STEPS

Insert Ethernet - IP Interface (RC/V 33.4)

Purpose

The Ethernet Internet Protocol (IP) Interface view (RC/V 33.4) provides the capability to provision an IP address, subnet mask, and associated IP parameters to an internet interface.

When to use

Use this procedure to define an Ethernet IP interface.

Repeat individual steps of this procedure as required.

Related information

Refer to 235-118-258, Recent Change Reference, for information on the RC/V system.

Admonishments

RC/V view 33.4 is Case Sensitive. Integrity of capital and lower case letters must be maintained.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release with the SIP for Packet Trunking - NAR feature software.

Required Information

- Switching Module Number
- PSU Number
- Shelf number
- Channel group number
- Gateway IP address(es)
- IP Subnet mask(s)

Required Conditions

- Protocol Handler is defined (RC/V 22.4)
- Channel Group is defined (RC/V 22.16)
- Internet Protocol (IP) Processor is assigned (RC/V 33.1)
- Processor Group is defined (RC/V 33.16)

Procedure

1 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

2 Type and enter the RC/V form number.

```
33.4
```

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

3 Type and enter the insert command.

```
I
```

Result:

Screen 1 of the ETHERNET INTERNET PROTOCOL (IP) INTERFACE ASSIGNMENT page is displayed and the cursor is located at the SM field.

4 Using screen 1 of the RC/V 33.4 form as a guide, type and enter the parameters.

- *SM - Switching Module number
- *PSU - Packet Switching Unit number
- *SHELF - PSU shelf number
- *CHANNEL GROUP - Channel group position
- #INTERFACE NAME - name of the interface (entries in this field are case sensitive)

5 Proceed to screen 2.

```
2
```

Result:

Screen 2 is displayed and the cursor is located at the GATEWAY IP ADDRESS 1 field.

6 On screen 2 of form 33.4, type and enter the parameters.

- #GATEWAY IP ADDRESS 1 - Primary IP address of the Ethernet-IP interface on the PH
- #IP SUBNET MASK 1 - subnetwork mask associated with the first gateway IP address
- GATEWAY IP ADDRESS 2- secondary IP address of the Ethernet-IP interface on the PH if the processor group to which the PH belongs will be supporting an SCTP near endpoint; blank if the PH will be supporting UDP paths
- IP SUBNET MASK 2 - subnetwork mask associated with the GATEWAY IP ADDRESS 2 (leave blank for UDP)
- MTU SIZE - 1500
- RATE - Rate associated with the speed of the Ethernet link
- MODE - mode used for data flow of the Ethernet connection

7 Type and enter the insert command.

`I`

Result:

`Inserting...form inserted`

NOTE: View MCC 118X,Y,X to see corresponding PH status change to Active.

8 Type and enter the previous screen command.

`<`

Result:

The INTERNET PROTOCOL VIEWS page is displayed.

9 If desired, type and enter the quit command to exit the RC/V system.

`Q`

Result:

The RC/V session is terminated.

10 Stop. You have completed this procedure.

END OF STEPS

Insert IP Routing (RC/V 33.3)

Purpose

The Internet Protocol (IP) Routing view (RC/V 33.3) provides the capability to provision an IP gateway between an external IP destination and a local IP interface.

When to use

Use this procedure to define the IP gateway between an external IP destination and a local IP interface.

Repeat individual steps of this procedure as required.

Related information

Refer to 235-118-258, Recent Change Reference, for information on the RC/V system.

Admonishments

RC/V view 33.3 is Case Sensitive. Integrity of capital and lower case letters must be maintained.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release with the SIP for Packet Trunking - NAR feature software.

Required Information

- Destination IP address.
- Gateway IP address

Required Conditions

- Protocol Handler is defined (RC/V 22.4)
- Channel Group is defined (RC/V 22.16)
- Internet Protocol (IP) Processor is assigned (RC/V 33.1)
- Processor Group is defined (RC/V 33.16)
- Ethernet IP interface is defined (RC/V 33.4)

Procedure

1 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

2 Type and enter the RC/V form number.

```
33.3
```

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

3 Type and enter the insert command.

```
I
```

Result:

The INTERNET PROTOCOL (IP) ROUTING TO INTERFACE page is displayed and the cursor is located at the DEST IP ADDR field

4 Using the RC/V 33.3 form as a guide, type and enter the parameters.

- *DEST IP ADDR - destination IP address that can be reached with this route
- *INTERFACE NAME - name of defined interface (entries in this field are case sensitive)
- NET OR HOST - NET (network host)
- IP SUBNET MASK - subnetwork mask associated with the destination IP address
- #GATEWAY IP ADDR - IP address of the gateway through which data is sent to the destination

5 Enter the insert command.

```
I
```

Result:

```
Inserting...form inserted
```

6 Type and enter the previous screen command.

```
<
```

Result:

The INTERNET PROTOCOL VIEWS page is displayed.

7 If desired, type and enter the quit command to exit the RC/V system.

`Q`

Result:

The RC/V session is terminated.

8 Stop. You have completed this procedure.

END OF STEPS

Insert Router Pinging (RC/V 33.17)

Purpose

The Internet Protocol (IP) Ping Parameters view (RC/V 33.17) defines pinging parameters pertaining to a router through a particular interface.

When to use

Use this procedure to define the pinging parameters of an access router.

Repeat individual steps of this procedure as required.

Related information

Refer to 235-118-258, Recent Change Reference, for information on the RC/V system.

Admonishments

RC/V view 33.17 is Case Sensitive. Integrity of capital and lower case letters must be maintained.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release with the SIP for Packet Trunking - NAR feature software.

Required Information

- IP Address of the adjacent router (use RC/V 33.3 - GATEWAY IP ADDR)
- Pinging interval
- Maximum number of failed pings

Required Conditions

- Protocol Handler is defined (RC/V 22.4)
- Channel Group is defined (RC/V 22.16)
- Secured Feature 684 is enabled (RC/V 8.22)
- SIP GSM is defined (RC/V 5.80)
- Processor Group is defined (RC/V 33.16)
- IP Processor interface is defined (RC/V 33.1)

- Ethernet IP interface is defined (RC/V 33.4)
- IP routing interface is defined (RC/V 33.3)

Procedure

1 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

2 Type and enter the RC/V form number.

```
33.17
```

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

3 Type and enter the insert command.

```
I
```

Result:

The INTERNET PROTOCOL (IP) PING PARAMETERS page is displayed and the cursor is located at the INTERFACE NAME field.

4 Using the RC/V 33.17 form as a guide, type and enter the parameters.

- *INTERFACE NAME - name identifying interface (entries in this field are case sensitive)
- *IP ADDRESS - address of the access router
- #PING - enabled or disabled (Y/N)
- PING INTVL - number of milliseconds between pings to the gateway router (increments of 100)
- MAX FAIL PINGS - maximum number of failed pings before pinging is terminated

5 Enter the insert command.

```
I
```

Result:

```
Inserting...form inserted
```

6 Type and enter the previous screen command.

`<`

Result:

The INTERNET PROTOCOL VIEWS page is displayed.

7 If desired, type and enter the quit command to exit the RC/V system.

`Q`

Result:

The RC/V session is terminated.

8 Stop. You have completed this procedure.

END OF STEPS

□

Insert UDP Path (RC/V 33.24)

Purpose

The UDP Path view (RC/V 33.24) defines UDP parameters pertaining to transport of SIP message.

When to use

Use this procedure to define a UDP path when SIP is to be supported by the UDP transport layer.

Repeat individual steps of this procedure as required.

Related information

Refer to 235-118-258, Recent Change Reference, for information on the RC/V system.

Admonishments

RC/V view 33.17 is Case Sensitive. Integrity of capital and lower case letters must be maintained.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR5 software release with UDP Transport for SIP software.

Required Information

- UDP Path number
- GSM number
- PSU number (0)
- Processor Group number
- Near-end UDP port
- Far-end IP address
- Far-end UDP port

Required Conditions

- Processor Group is defined (RC/V 33.16)
- Ethernet IP interface is defined (RC/V 33.4)
- IP routing exists for far-end IP address (RC/V 33.3), if far-end IP address is not in the same subnet as GATEWAY IP ADDR1 of the Ethernet/IP interface (RC/V 33.4)

Procedure

1 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

2 Type and enter the RC/V form number.

```
33.24
```

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

3 Type and enter the insert command.

```
I
```

Result:

The UDP Path page is displayed.

4 Using the RC/V 33.24 form as a guide, type and enter the parameters.

- UDP Path number
- GSM number
- PSU number (0)
- Processor Group number
- Near-end UDP port
- Far-end IP address
- Far-end UDP port

5 Enter the insert command.

```
I
```

Result:

```
Inserting...form inserted
```

6 Type and enter the previous screen command.

```
<
```

Result:

The INTERNET PROTOCOL VIEWS page is displayed.

7 If desired, type and enter the quit command to exit the RC/V system.

```
Q
```

Result:

The RC/V session is terminated.

8 Stop. You have completed this procedure.

END OF STEPS

☐

Insert SCTP Endpoint Timers and Protocol Parameters (RC/V 33.18)

Purpose

SCTP Endpoint Parameters (RC/V 33.18) defines the SCTP endpoint related timers and protocol parameters.

When to use

A default (DEFAULT) parameter set is defined when either the first SCTP near endpoint is provisioned or when the first non-DEFAULT SCTP endpoint parameter set is provisioned. This procedure should be followed to create a new parameter set, or to modify individual parameter values (within the allowable ranges for each parameter) that are different (non-default) from the default value.

Related information

Refer to 235-118-258, Recent Change Reference, for information on the RC/V system.

Admonishments

The parameters and timers in this form have default values. Changing any parameter or timer could cause an interruption in service. Care should be taken when changing the predefined values.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release with the SIP for Packet Trunking - NAR feature software.

Required Information

- SCTP Endpoint Parameter Set Name.

Required Conditions

- Secured Feature 684 is enabled (RC/V 8.22)
- SIP GSM is defined (RC/V 5.80)

Procedure

1 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

2 Type and enter the RC/V form number.

```
33.18
```

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

3 Type and enter the insert command.

```
I
```

Result:

The SCTP ENDPOINT PARAMETERS page is displayed and the cursor is located at the PARM SET NAME field.

4 Using the RC/V 33.18 form as a guide, type and enter the *PARM SET NAME.

5 Modify the following parameters as needed.

- RTO INI - Initial value of retransmission timer.
- RTO MIN - Minimum value of retransmission timer.
- RTO MAX - Maximum value of retransmission timer.
- RTO ALPHA - A constant used to estimate round-trip time.
- RTO BETA - A constant used to estimate retransmission time-out.
- VALID COOKIE LIFE - Lifetime of a cookie.
- HB INTRVL - Interval between transmitted heartbeat messages.
- BURST SIZE - A constant used to calculate how many bytes the sender is allowed to transmit.
- DELAY ACK TMR - This timer specifies how long the SCTP will wait before generating an SCTP packet specifically for a Selective Acknowledgement (SACK) chunk.
- FAST RETRANS CNT - Number of successive SACKs received before retransmitting a packet.

6 Enter the insert command.

`I`

Result:

`Inserting...form inserted`

7 Type and enter the previous screen command.

`<`

Result:

The INTERNET PROTOCOL VIEWS page is displayed.

8 If desired, type and enter the quit command to exit the RC/V system.

`Q`

Result:

The RC/V session is terminated.

9 Stop. You have completed this procedure.

END OF STEPS

□

Insert SCTP Near Endpoints (RC/V 33.19)

Purpose

The SCTP Endpoint (RC/V 33.19) defines the SCTP near endpoints to a processor group.

When to use

Use this procedure to define the near SCTP endpoints.

Repeat individual steps of this procedure as required.

Related information

Refer to 235-118-258, Recent Change Reference, for information on the RC/V system.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release with the SIP for Packet Trunking - NAR feature software.

Required Information

- Near EP name - Identifies the local SCTP endpoint.
- GSM - Global SM where processor group is located
- Processor Group Number

Required Conditions

- Secured Feature 684 is enabled (RC/V 8.22)
- SIP GSM is defined (RC/V 5.80)
- SCTP Parameters are defined if default is not acceptable (RC/V 33.18)
- Protocol Handler is defined (RC/V 22.4)
- Channel Group is defined (RC/V 22.16)
- Processor Group is defined (RC/V 33.16)
- Internet Protocol Processor is assigned (RC/V 33.1)
- Ethernet Internet Protocol Interface is defined (RC/V 33.4)

Procedure

1 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

2 Type and enter the RC/V form number.

```
33.19
```

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

3 Type and enter the insert command.

```
I
```

Result:

SCTP NEAR ENDPOINT DEFINITION page is displayed and the cursor is located at NEAR ENDPOINT NAME field.

4 Using the RC/V 33.19 form as a guide, type and enter the indicated values for each field.

- *NEAR ENDPOINT NAME
- #GSM
- #PCR GROUP
- SCTP PORT
- ENDPOINT PARM SET NAME - DEFAULT or customer defined

5 Enter the insert command.

```
I
```

Result:

```
Inserting...form inserted
```

6 Type and enter the previous screen command.

```
<
```

Result:

INTERNET PROTOCOL VIEWS page is displayed.

7 If desired, type and enter the quit command to exit the RC/V system.

`Q`

Result:

The RC/V session is terminated.

8 Stop. You have completed this procedure.

END OF STEPS

Insert SCTP Far Endpoints (RC/V 33.21)

Purpose SCTP Far Endpoint (RC/V 33.21) defines SCTP far endpoints.

When to use Use this procedure to define the far SCTP endpoints.

Repeat individual steps of this procedure as required.

Related information Refer to 235-118-258, Recent Change Reference, for information on the RC/V system.

Admonishments None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release with the SIP for Packet Trunking - NAR feature software.

Required Information

- Far endpoint name
- Destination SCTP port number
- IP address(es) of the far SCTP endpoint(s)

Required Conditions

- Protocol Handler is defined (RC/V 22.4)
- Channel Group is defined (RC/V 22.16)
- Secured Feature 684 is enabled (RC/V 8.22)
- SIP GSM defined (RC/V 5.80)
- Processor Group is defined (RC/V 33.16)
- Internet Protocol (IP) Processor is assigned (RC/V 33.1)
- Ethernet IP interface is defined (RC/V 33.4)
- Internet Protocol routing is defined (RC/V 33.3)

Procedure

1 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

2 Type and enter the RC/V form number.

`33.21`

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

3 Type and enter the insert command.

`I`

Result:

SCTP FAR ENDPOINT DEFINITION page is displayed and the cursor is located at FAR ENDPOINT NAME field.

4 Using the RC/V 33.21 form as a guide, type and enter the indicated values for each field.

- *FAR ENDPOINT NAME
- #DEST SCTP PORT
- #DEST IP ADDR 1
- DEST IP ADDR 2 - (optional)

5 Enter the insert command.

`I`

Result:

```
Inserting...form inserted
```

6 Type and enter the previous screen command.

`<`

Result:

INTERNET PROTOCOL VIEWS page is displayed.

7 If desired, type and enter the quit command to exit the RC/V system.

`Q`

Result:

The RC/V session is terminated.

8 Stop. You have completed this procedure.

END OF STEPS

Insert SCTP Association-related Protocol Parameters (RC/V 33.20)

Purpose

The SCTP Association Parameter Set (RC/V 33.20) view defines protocol parameters for an SCTP association.

When to use

This procedure defines protocol limits for an SCTP association. The DEFAULT association parameter set is created either when the first SCTP association is inserted or when the first non-DEFAULT association parameter set is inserted. This procedure should be followed to create a new parameter set, or to modify individual parameter values (within the allowable ranges for each parameter) that are different (non-default) from the default value.

Related information

Refer to 235-118-258, Recent Change Reference, for information on the RC/V system.

Admonishments

The parameters in this form have default values. Changing any parameter could cause an interruption in service. Care should be taken when changing the predefined values.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release with the SIP for Packet Trunking - NAR feature software.

Required Information

- None.

Required Conditions

- Secured Feature 684 is enabled (RC/V 8.22)
- SIP GSM is defined (RC/V 5.80)

Procedure

1 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

2 Type and enter the RC/V form number.

```
33.20
```

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

3 Type and enter the insert command.

```
I
```

Result:

The SCTP ASSOCIATION PARAMETER SET page is displayed and the cursor is located at the PARM SET NAME field.

4 Using the RC/V 33.20 form as a guide, type and enter the parameter set name in the *PARM SET NAME field.

5 Modify the parameters as needed.

- #STREAM NEGO - Indicates whether an association may be established if the desired number of outbound streams are not available.
- #PERMANENT - Indicates whether an association shall be maintained as a permanent connection or not.
- INBOUND STREAMS - Number of offered inbound streams.
- OUTBOUND STREAMS - Number of offered outbound streams.
- MAX PATH RETRANS - Maximum number of retransmissions of a datagram for an SCTP path before the path is declared out-of-service.

6 Enter the insert command.

```
I
```

Result:

```
Inserting...form inserted
```

7 Type and enter the previous screen command.

<

Result:

The INTERNET PROTOCOL VIEWS page is displayed.

8 If desired, type and enter the quit command to exit the RC/V system.

Q

Result:

The RC/V session is terminated.

9 Stop. You have completed this procedure.

END OF STEPS

Insert SCTP Association (RC/V 33.22)

Purpose

SCTP Association (RC/V 33.22) defines an association between near and far SCTP endpoints.

When to use

Use this procedure to create an association between a near and far SCTP endpoints.

Related information

Refer to 235-118-258, Recent Change Reference, for information on the RC/V system.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release with the SIP for Packet Trunking - NAR feature software.

Required Information

- Association Number - Identifies the SCTP association. This number must be unique for all SCTP associations provisioned in the office
- GSM - Identifies the Global SM that owns the processor group that supports the SCTP endpoint corresponding to this SCTP association
- Near endpoint
- Far endpoint.

Required Conditions

- Protocol Handler is defined (RC/V 22.4)
- Channel Group is defined (RC/V 22.16)
- Secured Feature 684 is enabled (RC/V 8.22)
- SIP GSM defined (RC/V 5.80)
- Processor Group is defined (RC/V 33.16)
- Internet Protocol (IP) Processor is assigned (RC/V 33.1)

- Ethernet IP interface is defined (RC/V 33.4)
- Internet Protocol routing is defined (RC/V 33.3)
- SCTP Parameters is defined (RC/V 33.18)
- SCTP near endpoint is defined (RC/V 33.19)
- SCTP Association Parameter Set is defined (RC/V 33.20)
- SCTP far endpoint is defined (RC/V 33.21)

Procedure

1 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

2 Type and enter the RC/V form number.

```
33.22
```

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

3 Type and enter the insert command.

```
I
```

Result:

SCTP ASSOCIATION page is displayed and the cursor is located at ASSOC NUMBER field

4 Using the RC/V 33.22 form as a guide, type and enter the indicated values for each field.

- *ASSOC NUMBER
- #GSM
- #NEAR ENDPOINT NAME
- #FAR ENDPOINT NAME
- SCTP ASSOC PARM SET NAME

5 Enter the insert command.

```
I
```

Result:

```
Inserting...form inserted
```

6 Type and enter the previous screen command.

<

Result:

INTERNET PROTOCOL VIEWS page is displayed.

7 If desired, type and enter the quit command to exit the RC/V system.

Q

Result:

The RC/V session is terminated.

8 Stop. You have completed this procedure.

END OF STEPS

Insert SCTP Association Set (RC/V 33.23)

Purpose

SCTP Association Set (RC/V 33.23) defines association sets between existing associations.

When to use

Use this procedure to define association sets.

Repeat individual steps of this procedure as required.

Related information

Refer to 235-118-258, Recent Change Reference, for information on the RC/V system.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release with the SIP for Packet Trunking - NAR feature software.

Required Information

- Association set name
- GSM number
- Association identification number(s)

Required Conditions

- Protocol Handler is defined (RC/V 22.4)
- Channel Group is defined (RC/V 22.16)
- Secured Feature 684 is enabled (RC/V 8.22)
- SIP GSM defined (RC/V 5.80)
- Processor Group is defined (RC/V 33.16)
- Internet Protocol (IP) Processor is assigned (RC/V 33.1)
- Ethernet IP Interface is defined (RC/V 33.4)
- Internet Protocol Routing is defined (RC/V 33.3)
- SCTP Parameters defined (RC/V 33.18)
- SCTP Near Endpoint defined (RC/V 33.19)

- SCTP Far Endpoint defined (RC/V 33.21)
- SCTP Association Parameter Set defined (RC/V 33.20)
- SCTP Association is defined (RC/V 33.22)

Procedure

1 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

2 Type and enter the RC/V form number.

```
33.23
```

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

3 Type and enter the insert command.

```
I
```

Result:

The ASSOCIATION SET page is displayed and the cursor is located at ASSOCIATION SET NAME field.

4 Using the RC/V 33.23 form as a guide, type and enter the indicated values for each field.

- *ASSOCIATION SET NAME (Recommend using the PKT GRP number in the Association Set name. This will aide trouble-shooting.)
- #GSM

5 Type and enter the unique association identification numbers that are in the association set in the NBR column. A maximum of 64 association ids can be defined in an association set. The associations must all belong to a specified SIP GSM.

6 Enter the insert command.

```
I
```

Result:

```
Inserting...form inserted
```

7 Type and enter the previous screen command.

<

Result:

The INTERNET PROTOCOL VIEWS page is displayed.

8 If desired, type and enter the quit command to exit the RC/V system.

```
Q
```

Result:

The RC/V session is terminated.

9 Proceed to MCC or STLWS to restore Association Set.

Restore the SCTP Association.

```
RST:SCTP,ASSOC=a;
```

Where:

a = Association number

Request the status of the Association:

```
OP:STATUS,SCTP,ASSOC,ALL;
```

Examine the output.

For additional information, refer to the SIP Maintenance Considerations procedure, *Resolve SCTP Association Problems,* 235-200-118.

10 Stop. You have completed this procedure.

END OF STEPS

Update CCS Office Parameters (RC/V 8.15)

Purpose The CCS Office Parameters view (8.15) contains the information that is used for Recent Change of Common Channel Signaling office parameter data.

When to use Use this procedure set the country code. The SIP country code is set to 1 for the United States, and to other values for different countries.

Related information Refer to 235-118-258, Recent Change Reference, for information on the RC/V system.

Admonishments None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- None.

Required Information

- Office identifier

Required Conditions

- Secured Feature 684 is enabled (RC/V 8.22)
- SIP GSM is defined (RC/V 5.80)

Procedure

1 Select and prepare terminal for RC/V activities.

Reference:

“Select and Prepare Terminal” (5-11)

2 Type and enter the RC/V form number.

```
8.15
```

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

3 Type and enter the review command.

`R`

Result:

The CCS OFFICE PARAMETERS page is displayed and the cursor is located at the OFFICE ID field.

4 Proceed to screen 2.

`2`

Result:

Screen 2 is displayed and the cursor is located at the bottom of Screen 2.

5 Verify that the *COUNTRY CODE field is set to 1.

NOTE: If it is not set to 1, enter the Update mode and type 1 in the *COUNTRY CODE field.

6 Proceed to screen X.

`X`

Result:

Screen X is displayed and the cursor is located at the bottom of Screen X.

7 Specify the NATIONAL CIRCUIT CODE field.

8 Enter the update command.

`U`

Result:

`Updating...form updated`

9 Type and enter the previous screen command.

`<`

Result:

Screen 1 of the OFFICE MISC. & ALARMS VIEWS page is displayed.

10 If desired, type and enter the quit command to exit the RC/V system.

`Q`

Result:

The RC/V session is terminated.

11 Stop. You have completed this procedure.

END OF STEPS

Insert SIP PSTN Interworking Parameter Set (RC/V 5.83)

Purpose

The SIP PSTN Interworking Parameter Set view (RC/V 5.83) is used to define interworking between SIP and ISUP parameters. The new view consists of two screens; one for PSTN to SIP interworking and a second for mapping from SIP to PSTN.

When to use

Use this procedure to create a new SIP PSTN interworking parameter set to use instead of the DEFAULT parameter sets. The DEFAULT SIP PSTN interworking parameter sets, “DEFAULTISUP” and “DEFAULTNOISUP”, are created when the first SIP GSM is inserted. This procedure should also be used to modify individual parameter values (within the allowable ranges for each parameter) that are different (non-default) from the default value.

The RC/V view 5.71 uses the set name to associate it with a packet group.

Related information

Refer to 235-118-258, Recent Change Reference, for information on the RC/V system.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS®* switch must be running on the 5E16.2 FR6 or later software release with the SIP for Packet Trunking - NAR feature software.

Required Information

- SIP PSTN interworking parameter set name

Required Conditions

- Secured Feature 684 is enabled (RC/V 8.22)
- SIP GSM is defined (RC/V 5.80)

Procedure

1 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

2 Type and enter the RC/V form number.

```
5.83
```

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

3 Type and enter the insert command.

```
I
```

Result:

The SIP PSTN INTERWORKING PARAMETER SET page is displayed and the cursor is located at the SET NAME field.

4 Type and enter the *SET NAME field value.

5 Using the RC/V 5.83 form as a guide, modify the parameters if necessary.

The fields listed below appear on Screen 1 and are the SIP TO PSTN MAPPING RULES.

- CALLING PARTY INFO
- CALLED PARTY NUMBER
- TRANSIT NETWORK SELECTION
- REDIRECTING INFO
- ORIGINATING LINE INFO
- HOP COUNTER
- CARRIER SELECTION

The fields listed below appear on Screen 2 and are the PSTN TO SIP MAPPING RULES.

- REQUEST URI ADDR
- REQUEST URI CIC

- REQUEST URI CSEL
- TO HEADER
- FROM ADDR
- FROM NAME
- FROM OLI
- P-ASSERTED ADDR
- P-ASSERTED NAME
- DIVERSION
- MAX FORWARDS

6 Enter the insert command.

`I`

Result:

`Inserting...form inserted`

A list of near endpoints is displayed.

7 Type and enter the previous screen command.

`<`

Result:

The TRUNK VIEWS page is displayed.

8 If desired, type and enter the quit command to exit the RC/V system.

`Q`

Result:

The RC/V session is terminated.

9 Stop. You have completed this procedure.

END OF STEPS

Table 5-1 SIP to PSTN Mapping Reference

Field	Value	Result
CALLED PARTY NUMBER	ISUPMINE	Received SIP INVITEs ISUP MINE's Called Party Number parameter and portability are derived from the ISUP MIME data. This includes Called Number, Ported Number GAP, Ported Number Translation Indicator. ***Note:*** When this is set, an ISUP MINE is mandatory in the initial INVITE message.
	BASIC	Received SIP INVITEs Request URI provides Called Party Number and portability data. The Called Party Number comes from the Request URI xyz portion; Ported Number GAP from request URI xyz portion; Ported Number Translation Indicator from the Request URI ″rn″ tag.

Table 5-2 PSTN to SIP Mapping Reference

Field	Value	Result
Request URI	BASIC	The received SIP message's must contain an ISUP MIME with a Called Part.
Request URI CSEL	BASIC	″csel″ tag is appended to the INVITE's ″Request URI″ when available from a carrier call.
	NOTMAPPED	″csel″ tag is not sent in the outgoing SIP INVITE headers.

□

Insert SIP Parameters (RC/V 5.82)

Purpose

The SIP-T Parameters Definition view (RC/V 5.82) is used to define SIP parameter set for the packet trunking application.

When to use

Use this procedure to create a new SIP parameter set to use instead of the DEFAULT parameter set. The DEFAULT SIP parameter set is created when the first SIP GSM is inserted. This procedure should also be used to modify individual parameter values (within the allowable ranges for each parameter) that are different (non-default) from the default value.

Related information

Refer to 235-118-258, Recent Change Reference, for information on the RC/V system.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running on the 5E16.2 FR3 or later software release with the SIP for Packet Trunking - NAR feature software.

Required Information

- None.

Required Conditions

- Secured Feature 684 is enabled (RC/V 8.22)
- SIP GSM is defined (RC/V 5.80)
- Secured Feature 769 is enabled (RC/V 8.22) if SIP without Encapsulated ISUP is used.

Procedure

1 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

---

2 Type and enter the RC/V form number.

```
5.82
```

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

---

3 Type and enter the insert command.

```
I
```

Result:

The SIP-T PARAMETERS DEFINITION page is displayed and the cursor is located at the PARM SET NAME field.

---

4 Type and enter the *PARM SET NAME field value.

---

5 Using the RC/V 5.82 form as a guide, modify the timers and parameters if necessary.

- INVITE TMR
- NON INVITE TMR
- WAIT FOR ACK
- WAIT FOR UPDATE
- INIT RETR TMR
- AUTOMATIC ACM TMR
- SIG SERVICE
- SERVICE CDPT
- TRUST ISUP
- MSG BUNDLE DELAY
- ISUP ENCAPSULATION
- PRACK ENABLED
- WAIT FOR PRACK

NOTE: The service provider is responsible for ensuring proper coordination of its switches. If SIP without Encapsulated ISUP is being provisioned, the packet group at the TPS which will receive the INVITE without ISUP must have TRUST ISUP set to **no**. Trust ISUP=N is the proper setting when connecting a TPS to a proxy and there are multiple network elements on the other side with varying capabilities. Trust ISUP=Y is the proper setting when two 'true' switches that understand ISUP are directly connected.

NOTE: The service provider is responsible for ensuring proper coordination of its switches. If SIP without Encapsulated ISUP is being provisioned, the packet group at the TPS which will receive the INVITE without ISUP must have an appropriate SIP to PSTN Interworking Parameter set (RC/V View 5.83) assigned. Refer to the *Feature Description,* 235-190-400, document for SIP to PSTN Interworking Parameter set guidelines.

6 Enter the insert command.

`I`

Result:

`Inserting...form inserted`

A list of near endpoints is displayed.

7 Type and enter the previous screen command.

`<`

Result:

The TRUNK VIEWS page is displayed.

8 If desired, type and enter the quit command to exit the RC/V system.

`Q`

Result:

The RC/V session is terminated.

9 Stop. You have completed this procedure.

END OF STEPS

Insert Packet Group (RC/V 5.71)

Purpose

The Packet Group view (RC/V 5.71) is used to insert a SIP packet group. A SIP packet group is used to designate packet trunking to a given office.

When to use

Use this procedure to add a SIP packet group.

Related information

Refer to 235-118-258, Recent Change Reference, for information on the RC/V system.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release with the SIP for Packet Trunking - NAR feature software.

Required Information

- Packet Group number
- Bearer network ID
- Far end CLLI
- Near end CLLI
- Hunt type
- Control type
- Association set name or UDP path number
- SIP GSM number
- SIP parameter set name
- SIP PSTN interworking parameter set name
- Preconditioning
- Local audible ringing

Required Conditions

- SIP GSM is defined (RC/V 5.80)
- Association set is defined (RC/V 33.23) or UDP path is defined (RC/V 33.24)

Procedure

1 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

2 Type and enter the RC/V form number.

```
5.71
```

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

3 Type and enter the insert command.

```
I
```

Result:

The PACKET GROUP DEFINITION page is displayed and the cursor is located at the PKT GRP field.

4 Using the RC/V 5.71 form as a guide, type and enter the parameters.

- *PKT GRP - packet group number
- #BEARER NET ID - bearer network identifier
- #FAR CLLI - Far end Common Language Location Identifier
- #NEAR CLLI - Near end Common Language Location Identifier
- #HUNT TYPE - 2WF
- #CONTROL TYPE - CANCEL
- One of the following:
  - ASSOC SET NAME - Association set name, or
  - UDP PATH - UDP path number.
- SIP-T GSM - GSM Number
- SIP-T PARM SET NAME - SIP parameter set name

- SIP PSTN SET NAME - SIP PSTN interworking parameter set name
- SIP-T PRECOND - Y or N for Association set, N for UDP path
- LOCAL AUDIBLE - whether or not local audible ringing can be provided

---

5 Enter the insert command.

`I`

Result:

`Inserting...form inserted`

---

6 Type and enter the previous screen command.

`<`

Result:

The TRUNK VIEWS page is displayed.

---

7 If desired, type and enter the quit command to exit the RC/V system.

`Q`

Result:

The RC/V session is terminated.

---

8 Stop. You have completed this procedure.

END OF STEPS

Assign Trunk Group (RC/V 5.1)

Purpose
The Trunk Group view (RC/V 5.1) is used to assign trunk groups.

When to use
Use this procedure to define a trunk group to be used for packet trunking.

Related information
Refer to 235-118-258, Recent Change Reference, for information on the RC/V system.

Admonishments
None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release with the SIP for Packet Trunking - NAR feature software.

Required Information

- None.

Required Conditions

- PKT GRP is defined (RC/V 5.71)

Procedure

1 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

2 Type and enter the RC/V form number.

```
5.1
```

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

3 Type and enter the insert command.

`I`

Result:

Screen 1 of the TRUNK GROUP page is displayed.

4 Enter the appropriate Trunk Group number in the TGN field.

Result:

The TRUNK GROUP page data is displayed.

5 Enter the appropriate mandatory field.

- FAR CLLI match far clli on 5.71
- TRK DIR twoway
- HUNT TYP 2wf
- SCR
- GLSRE ACTION oddeven
- DAS
- TRK CLASS ic|lata|ltollcon|ttollcon
- INPLS isup7
- OUTPLS isup7
- FAR END NPA
- MODULE 0(5ESS) SM# (DRM/VCDX)
- NEAR CLLI match nearclli on 5.71
- CONTROL TYPE cancel
- CCS7 TYPE

6 Proceed to screen 12.

`12`

Result:

Screen 12 of the TRUNK GROUP page is displayed and the cursor is at the bottom of Screen 12.

7 Using the RC/V 5.1 form as a guide, type and enter the parameters under the PACKET NETWORK TRUNKING column.

- PKT GRP
- SIPT GRP - Y
- DIFF SERV

8 Type and enter the insert command.

`I`

Result:

`Inserting...form inserted`

9 Type and enter the previous screen command.

<

Result:

TRUNKS VIEWS page is displayed.

10 If desired, type and enter the quit command to exit the RC/V system.

`Q`

Result:

The RC/V session is terminated.

11 Stop. You have completed this procedure.

END OF STEPS

Enable INVITE Requests (RC/V 5.81)

Purpose

The SIP-T Call Processing SMs view (RC/V 5.81) defines SM-2000s that can serve as a call processing SM for SIP calls.

When to use

Use this procedure to disable or enable an SM-2000 from processing SIP calls.

Related information

Refer to 235-118-258, Recent Change Reference, for information on the RC/V system.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release with the SIP for Packet Trunking - NAR feature software.

Required Information

- SM-2000 module

Required Conditions

- SIP GSM is provisioned (RC/V 5.80)
- Protocol Handler is defined (RC/V 22.4)
- Channel groups are provisioned (RC/V 22.16)
- QPH pipes are provisioned (RC/V 17.24)
- MH pipes are provisioned (RC/V 17.20)
- Global SM/GQPH connectivity is provisioned (RC/V 17.27)

NOTE: GQPH pipes and QLPS connectivity are not required for DRM/VCDX .

Procedure

1 Type and enter the RC/V form number.

`5.81`

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

2 Type and enter the insert command.

```
U
```

Result:

The SIP-T CALL PROCESSING SMS page is displayed and the cursor is located at the GSM field.

3 Using the RC/V 5.81 form as a guide, type and enter the parameters.

- * GSM
- SM2K MODULE - number
- ROUTE - Y

4 Enter the insert command.

```
U
```

Result:

```
Updating...form updated
```

NOTE: If GSM# equals SM2K#, a Warning message is displayed.

“`SM2K Module in SIP-T SMS list should not equal GSM. This is not applicable for DRM/VCDX office.`”

If GSM should/does equal non-GSM enter “I” to ignore the warning and continue.

5 Type and enter the previous screen command.

```
<
```

Result:

The TRUNKS VIEWS page is displayed.

6 If desired, type and enter the quit command to exit the RC/V system.

```
Q
```

Result:

The RC/V session is terminated.

7 Stop. You have completed this procedure.

END OF STEPS

Assign Route Index (RC/V 10.2)

Purpose

The Route Index view (RC/V 10.2) is used to create route indices. A route index determines a network path, typically used to complete a call or route to error treatment when the call is unable to complete.

When to use

Use this procedure to assign a Trunk Group to a route.

Related information

Refer to 235-118-258, Recent Change Reference, for information on the RC/V system.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release with the SIP for Packet Trunking - NAR feature software.

Required Information

- Route Index - number used for all call terminating to a telephone number.
- Trunk Group Number - TGN defined in (RC/V 5.1)

Required Conditions

- None.

Procedure

1 Select and prepare terminal for RC/V activities.

Reference:

“Select and Prepare Terminal” (5-11)

2 Type and enter the RC/V form number.

`10.2`

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
  D=Delete:
```

3 Type and enter the insert command.

`I`

Result:

Screen 1 of the ROUTE INDEX (ROUTING) page is displayed and the cursor is at the RTI field.

4 Using the RC/V 10.2 form as a guide, type and enter the parameters.

- *RTI - route index (1-16382)
- TGN - trunk group number (0-4000)

5 Type and enter the insert command.

`I`

Result:

```
Inserting...form inserted
```

6 Type and enter the previous screen command.

`<`

Result:

ROUTING & CHARGING VIEWS page is displayed.

7 If desired, type and enter the quit command to exit the RC/V system.

`Q`

Result:

The RC/V session is terminated.

8 Stop. You have completed this procedure.

END OF STEPS

Add/Change SIP Elements within an Existing SIP Network

Introduction

This section contains additional procedures that might be required for Provisioning the SIP signaling network.

List of procedures

Below are additional steps for provisioning the SIP application. This section assumes the reader is knowledgeable about the RC/V system and the reading of office records. More procedural information can be found in the links that are listed within the steps.

1. “Add New SIPT PHE2 IP Processor Group on an Existing SIP GSM” (5-95)
2. “Add New SCTP Near Endpoint” (5-99)
3. “Add New SCTP Association to Connect to a SCTP Far Endpoint” (5-102)
4. “Add New Association Set to Another Office” (5-105)
5. “Add SCTP Associations to Existing Association Set” (5-107)
6. “Add UDP Transport for SIP Signaling to Another Office” (5-109)
7. “Add New SIP Packet Trunking to Another Office” (5-112)
8. “Add New SIP Call-Processing SMs” (5-116)
9. “Add a SIP PHE2 to Existing Simplex SIP IP Processor Group” (5-118)
10. “Change IP/SCTP Transport Address for SCTP Near Endpoint” (5-121)
11. “Change IP/SCTP Transport Address for SCTP Far Endpoint” (5-126)
12. “Change Far IP Transport Address for UDP Path” (5-129)
13. “Change IP Address for Adjacent IP Router” (5-132)
14. “Change IP Parameters for a SIP processor group” (5-136)
15. “Change SCTP Endpoint Parameters” (5-142)
16. “Change SCTP Association Parameters” (5-144)
17. “Change SIP Parameters” (5-147)
18. “Provision Alarm Level for IP Fragmented Packets Beyond PM Threshold” (5-150)

    This procedure is executed to enable the IP Fragmented Packets Alarm. By default the alarm is disabled.

19. “Provision Alarm Level for ICMP Echo Requests Beyond PM Threshold” (5-153)
    This procedure is executed to enable the ICMP Echo Request Alarm. By default the alarm is disabled.
20. “Change SIP PSTN Interworking Parameter Set” (5-156)

□

Add New SIPT PHE2 IP Processor Group on an Existing SIP GSM

Purpose

This procedure defines the procedure for adding a new SIP PHE2 IP processor group on an existing SIP GSM.

When to use

Use this procedure to add a new SIP PHE2 IP processor group on an existing SIP GSM.

Related information

Refer to 235-118-258, Recent Change Reference, for information on the RC/V system.

Refer to 235-080-100, Translations Guide, for information on Office Records (output reports).

Refer to the 235-600-750, Output Messages manuals for additional information on the Output commands.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release with the SIP for Packet Trunking - NAR feature software.

Required Information

- SIP GSM Number
- PSU Shelf and position information

Required Conditions

- Available slot to install PHE2 board(s) on PSU shelf(s) with a DF2.
- Available grounding plate for installing 100BaseT Ethernet link(s)

Procedure

1 Execute the Hardware Growth Procedures to install the desired number of PHE2s and Ethernet Links (1 or 2, for Simplex or Duplex).

Each PHE2 should be in the STBY state on the 118X page for the PSU shelf on which it resides.

2 Use the procedure "PH Channel Group Assignment (RC/V 22.16)" (5-19) to update the PH TYPE of the desired channel group number(s) from NULL to PHE2 for the PSU shelf where each PHE2 was installed.

3 Use the procedure "Assign PH Channel Group as IP Processor (RC/V 33.1)" (5-34) to insert each channel group as an IP Processor, with the MTU ENABLE set to Y, and the IP Parameters identical for both channel groups that will be used for a duplex processor group.

4 Dump the ETHIP/5995 office records. At the MCC or TLWS type and enter the command:

```
OP:OFR:FORM=5995;
```

Result:

```
The ETHIP/5995 office records are dumped.
```

5 Identify the Ethernet Link numbers that are in use on the existing SIP GSM and select unused PSELNK number(s) for the new PHE2 Ethernet Link(s).

6 Dump the PCRGRP/5883 office records. At the MCC or TLWS type and enter the command:

```
OP:OFR:FORM=5883
```

Result:

```
OP:OFR:FORM=5883 PF is printed followed by a printout
    of the PCRGRP/5883 office records.
```

7 Identify the processor group numbers that are in use on the existing SIP GSM and select unused PCRGRP number(s) for the new processor group(s).

8 Use the procedure “Insert Processor Group (RC/V 33.16)” (5-38) to insert the PCRGRP using the PHE2 positions, channel group numbers, Ethernet Link numbers, and processor group numbers identified above.

9 At the MCC or TLWS type and enter the backup command.

```
BKUP:ODD;
```

Result:

```
BKUP ODD   COMPLETED
```

10 On the 118X page, the PHE2s should now be DGR.

11 Verify the IP addresses of the adjacent router(s), and select appropriate IP address(es) for the new endpoint.

Some things to consider:

- The IP address(es) selected must be unique within the switch and the external IP network it will be connected to.
- When assigning 2 IP addresses, for the additional reliability of a multi-homed endpoint, reliability can be gained by ensuring that the 2 IP addresses are in different IP subnets.
- The IP address(es) of the adjacent IP router(s) to which the Ethernet link(s) are connected must be in the same IP subnet as at least one of the IP addresses selected for the Ethernet IP interface.

12 On RC/V 33.4, for one of the channel groups in the processor group that will support the endpoint, insert the new Ethernet IP interface, including IP interface name, IP addresses and subnet masks. Set the MTU SIZE to 1500, required for SIP-T.

Reference:

“Insert Ethernet - IP Interface (RC/V 33.4)” (5-41)

13 On the 118X page, the PHE2(s) should now be ACT.

14 Stop. You have completed this procedure.

END OF STEPS

Add New SCTP Near Endpoint

Purpose

This procedure defines the steps for adding a new SCTP near endpoint.

When to use

Use this procedure to add a new SCTP near endpoint.

Related information

Refer to 235-118-258, Recent Change Reference, for information on the RC/V system.

Refer to 235-080-100, Translations Guide, for information on Office Records (output reports).

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release with the SIP for Packet Trunking - NAR feature software.

Required Information

- IP Addresses of adjacent routers

Required Conditions

- None

Procedure

1 Examine the PCRGRP/5883 and SCTPNEPD/5886 office records to identify a processor group that does not have an SCTP near endpoint assigned to it or execute the "Add New SIPT PHE2 IP Processor Group on an Existing SIP GSM" (5-95) procedure.

2 If there are two PHE2s in the processor group, read RC/V 33.4 for the other channel group, to make sure identical IP interface data was automatically inserted for it.

3 On RC/V 33.3, provision the IP gateway between an external IP destination and a local IP interface.

Reference:

"Insert IP Routing (RC/V 33.3)" (5-44)

4 On RC/V 33.17, use the new IP interface name and each adjacent router IP address to insert IP ping parameters from the endpoint to the router(s).

Reference:

"Insert Router Pinging (RC/V 33.17)" (5-47)

5 Use EXC:PING, OP:TCPIP:ARPDMP, and OP:TCPIP:RTDMP to verify reachability of the adjacent IP router from new Ethernet IP interface.

6 Examine the SCTPPARM/5885 office records and select an appropriate SCTP endpoint parameter set, or do a change/insert operation on RC/V 33.18 for an existing parameter set to add a new parameter set.

7 On RC/V 33.19, insert the new SCTP Near Endpoint, using the selected processor group and SCTP parameter set.

Reference:

"Insert SCTP Near Endpoints (RC/V 33.19)" (5-56)

8 Type and enter the backup command .

```
BKUP:ODD;
```

Result:

```
BKUP ODD   COMPLETED
```

9 Use OP:STATUS,SCTP,NEAREPT to verify that the status of the endpoint just grown is GROW.

10 Stop. You have completed this procedure.

END OF STEPS

Add New SCTP Association to Connect to a SCTP Far Endpoint

Purpose This procedure contains the steps for adding a new SCTP association to connect to a SCTP far endpoint.

When to use Use this procedure to connect a new SCTP association to a SCTP far endpoint.

Related information Refer to 235-118-258, Recent Change Reference, for information on the RC/V system.

Refer to 235-080-100, Translations Guide, for information on Office Records (output reports).

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Admonishments None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release with the SIP for Packet Trunking - NAR feature software.

Required Information

- Existing SCTP Far Endpoint name, or IP addresses and SCTP port to insert new SCTP Far Endpoint

Required Conditions

- None

Procedure

1 Examine the SCTPNEPD/5886 office records to select an existing SCTP near endpoint for the association, or execute the “Add New SCTP Near Endpoint” (5-99)” procedure.

---

2 Examine the ASSCPARM/5887 office records and select an appropriate SCTP association parameter set, or do a change/insert operation on RC/V 33.20 for an existing parameter set to add a new parameter set.

---

3 If the association is to be added to an SCTP Far Endpoint that is already associated with other SCTP Near Endpoints in the *5ESS*® switch, proceed to step 4.

Otherwise:

- Obtain the IP addresses and SCTP port of the new far endpoint, and use OP:TCPIP:RTDMP and/or examine the IPROUT/5989 office records to verify whether the necessary IP route(s) exist to the far IP address(es) from the near IP interface.
- If the required IP route(s) do not exist, use RC/V 33.3 to insert them. Refer to the "Insert IP Routing (RC/V 33.3)" (5-44) procedure.
- Use EXC:PING and OP:TCPIP:RTDMP to verify the reachability of the new IP addresses from the near IP interface.
- On RC/V 33.21, insert the new SCTP far endpoint. Refer to the "Insert SCTP Far Endpoints (RC/V 33.21)" (5-59) procedure.

---

4 Examine the SCTPASSC/5889 office records to select an unused association number.

---

5 On RC/V 33.22, use the selected association number to insert the association, with the selected near endpoint, far endpoint, and association parameter set.

- If this was the first association on an SCTP near endpoint that was in the GROW state, there should be an autonomous report that the endpoint has transitioned from GROW to NOSERV.
- If this was not the first association, there may be a report that the endpoint has transitioned from INSERV to PARTSERV, if all the previously-existing associations are ESTABLISHED.

---

6 At the MCC or TLWS type and enter the backup command.

```
BKUP:ODD;
```

Result:

```
BKUP ODD   COMPLETED
```

7 Use OP:STATUS,SCTP,ASSOC to check the status of the new association. It should be in the CLOSED MAN state.

It cannot be established until it has been added to an association set, manually restored, and the far endpoint is in an INSERV state.

8 Stop. You have completed this procedure.

END OF STEPS

Add New Association Set to Another Office

Purpose

This procedure contains the steps for adding a new association set to another office.

When to use

Use this procedure to add a new association set to another office.

Related information

Refer to 235-118-258, Recent Change Reference, for information on the RC/V system.

Refer to 235-080-100, Translations Guide, for information on Office Records (output reports).

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release with the SIP for Packet Trunking - NAR feature software.

Required Information

- None

Required Conditions

- None

Procedure

1 Examine the SCTPASSC/5889 office records to identify the associations to be used in the new association set, or execute the procedure “Add New Association Set to Another Office” (5-105) for each association that will connect an SCTP near endpoint in the *5ESS*® switch to an SCTP far endpoint in the other office.

Note: All the SCTP near endpoints of the associations on the set must be on the same SIP-T GSM. The near endpoints on one SIP-T GSM can be determined from RC/V 5.80, SIP GSM.

2 On RC/V 33.23, insert the new association set on the GSM.

Reference:

" Insert SCTP Association Set (RC/V 33.23)" (5-68)

3 At the MCC or TLWS type and enter the backup command.

```
BKUP:ODD;
```

Result:

```
BKUP ODD   COMPLETED
```

4 Verify that corresponding provisioning of all associations in the set is complete at the far office.

If the far office is a *5ESS*® switch, also verify that the provisioning of the association set at the other office is complete and consistent with the provisioning at the near end.

5 Use the RST:SCTP,ASSOC command to establish the associations.

The far office must also be attempting to establish the associations. Use OP:STATUS,SCTP,ASSOCSET to verify that the associations have been established.

6 Stop. You have completed this procedure.

END OF STEPS

Add SCTP Associations to Existing Association Set

Purpose

This section defines the procedure for adding an SCTP association to an existing association set.

When to use

Use this procedure to add an SCTP association to an existing association set.

Related information

Refer to 235-118-258, Recent Change Reference, for information on the RC/V system.

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release with the SIP for Packet Trunking - NAR feature software.

Required Information

- Association number
- Packet Group

Required Conditions

- None

Procedure

1 Use OP:STATUS,SCTP,ASSOC and/or read RC/V 33.22 for the association to verify that it is not already part of an association set, and is in the CLOSED MAN state, or execute the "Add New SCTP Association to Connect to a SCTP Far Endpoint" (5-102) procedure.

2 Use RC/V 33.23 to add the association to the association set.

Reference:

[“ Insert SCTP Association Set (RC/V 33.23)” (5-68)]

---

3 At the MCC or TLWS type and enter the backup command.

```
BKUP:ODD;
```

Result:

```
BKUP ODD   COMPLETED
```

---

4 Verify that equivalent provisioning of the association has been done at the far endpoint.

---

5 Use RST:SCTP,ASSOC to restore the association and OP:STATUS,SCTP,ASSOC to verify that it becomes ESTABLISHED.

---

6 Stop. You have completed this procedure.

END OF STEPS

Add UDP Transport for SIP Signaling to Another Office

Purpose

This section defines the procedure for making UDP transport available for SIP signaling to another office.

When to use

Use this procedure when SIP signaling to a new far office is to be supported by UDP transport

Related information

Refer to 235-118-258, Recent Change Reference, for information on the RC/V system.

Refer to 235-080-100, Translations Guide, for information on Office Records (output reports).

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR5 software release with UDP Transport for SIP software.
- If the new SIP Packet Group is to be supported by UDP transport and/or requires SIP without Precondition Procedures, the *5E-XC*™ must be running a 5E16.2 FR5 or later software release that supports those features.

Required Information

- The UDP port number at the near office, and the IP address and UDP port number at the far office for SIP signaling.

Required Conditions

- SIP GSM is provisioned

Procedure

1 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

---

2 Verify that SFID 700 is enabled, or execute the Feature Activation (RC/V 8.22) procedure to enable it.

---

3 Examine the PCRGRP/5883 office records to determine the SIP GSM and provisioned SIP processor groups. If SCTP transport exists in the office, examine the SCTPNEPD/5886 office records to exclude any processor group that has an SCTP endpoint assigned to it. If there are no available processor groups without SCTP endpoints, execute the "Add New SIPT PHE2 IP Processor Group on an Existing SIP GSM" (5-95) procedure. Otherwise, an existing SIPT PHE2 IP Processor Group may be selected, since multiple UDP paths may terminate on the same processor group

---

4 If the far IP address is not in the same IP subnet as the IP address of the Ethernet-IP interface of the PHs in the selected processor group:

- use `OP:TCPIP,RTDMP` and/or
- examine the IPROUT/5989 office records to verify that the necessary IP route is provisioned, or

---

5 execute the "Insert IP Routing (RC/V 33.3)" procedure.

Reference:

"Insert IP Routing (RC/V 33.3)" (5-44)

---

6 use `EXC:PING` to verify the reachability of the new far IP address from the near IP interface.

---

7 examine the UDPPATH/5891 office records to select an unused UDP Path number.

---

8 Execute the "Insert UDP Path (RC/V 33.24)" procedure.

Reference:

"Insert UDP Path (RC/V 33.24)" (5-50)

9 At the MCC or TLWS type and enter the backup command.

```
BKUP:ODD;
```

Result:

```
BKUP ODD   COMPLETED
```

10 Stop. You have completed this procedure.

END OF STEPS

Add New SIP Packet Trunking to Another Office

Purpose

This section defines the procedure for adding a SIP packet trunking to another office.

When to use

Use this procedure to add SIP packet trunking to another office

Related information

Refer to 235-118-258, Recent Change Reference, for information on the RC/V system.

Refer to 235-080-100, Translations Guide, for information on Office Records (output reports).

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release with the SIP for Packet Trunking - NAR feature software.
- If the new SIP Packet Group is to be supported by UDP transport and/or requires SIP without Precondition Procedures, the *5E-XC*™ must be running a 5E16.2 FR5 or later software release that supports those features.

Required Information

- The CLI of the far office.
- The type of transport (SCTP versus UDP) to be used for signaling to the far office.
- Whether or not the far office supports the Precondition procedures for SIP.
- The Bearer Network ID of the OIU-IP(s) to be used for the bearer to the far office.

- Which SIP PSTN parameter set should be used, if any.
- Whether local audible ringing should be supplied.

Required Conditions

- None

Procedure

1 Select and prepare terminal for RC/V activities.

Reference:

“Select and Prepare Terminal” (5-11)

2 If SCTP transport is to be used for SIP signaling to the other office, examine the ASSCSET/5890 office records to identify the association set to be used, or execute the “Add New Association Set to Another Office” (5-105) procedure.

3 If UDP transport is to be used:

- for SIP signaling to the other office, verify that SFID 700 is enabled, or execute the Feature Activation (RC/V 8.22) procedure.
- for SIP signaling, examine the UDPPATH/5891 office records to identify the UDP path to be used, or execute the “Add UDP Transport for SIP signaling to Another Office” procedure.

4 Examine the SIPTPARM/5882 office records and select an appropriate SIP-T parameter set, or do a change/insert operation on RC/V 5.82 for an existing parameter set to add a new parameter set.

Reference:

“Insert SIP Parameters (RC/V 5.82)” (5-79)

5 Examine the SIPPSTN/5893 office records and select an appropriate SIP PSTN interworking parameter set, or do a change/insert operation on RC/V 5.83 for an existing parameter set to add a new parameter set.

Reference:

"Insert SIP PSTN Interworking Parameter Set (RC/V 5.83)" (5-74)

6 Examine the PKTGRP/5219 office records to select an unused packet group number for the new packet group.

7 On RC/V 5.71, insert the packet group, using the information determined in the previous steps.

Reference:

"Insert Packet Group (RC/V 5.71)" (5-82)

8 Examine the TKGRP/5202 office records to select an unused trunk group number for the new packet group.

9 On RC/V 5.1, insert the new trunk group for the new packet group, with SIPT GRP set to Y.

Reference:

"Assign Trunk Group (RC/V 5.1)" (5-85)

10 At the MCC or TLWS type and enter the backup command.

```
BKUP:ODD;
```

Result:

```
BKUP ODD   COMPLETED
```

11 Insert screening/routing/digit analysis data to allow outgoing calls to be routed to the trunk group provisioned in this procedure.

Reference:

"Assign Route Index (RC/V 10.2)" (5-91)

12 Make test calls on the new packet group/trunk group using TST:PATH:OG105 command

13 At the MCC or TLWS type and enter the backup command.

`BKUP:ODD;`

Result:

```
BKUP ODD   COMPLETED
```

14 Stop. You have completed this procedure.

END OF STEPS

Add New SIP Call-Processing SMs

Purpose

This section contains the procedure for adding a new SIP call-processing SM.

When to use

Use this procedure to add a new SIP call-processing SM.

Related information

Refer to 235-118-258, Recent Change Reference, for information on the RC/V system.

Refer to 235-080-100, Translations Guide, for information on Office Records (output reports).

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release with the SIP for Packet Trunking - NAR feature software.

Required Information

- SIP GSM numbers
- CP SM number(s)

Required Conditions

- None

Procedure

1 Use OP:STATUS,GQPHLNK or examine the QGCON/5829 office records to verify that connectivity exists between the desired new SIP CP SM(s) and the SIP-T global SM/PSU.

2 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

3 Update RC/V 17.27 to add any necessary connectivity.

Reference:

"Update GSM - Non-GSM Communication (RC/V 17.27)" (5-28)

4 Update RC/V 5.81 to add the new SIP CP SMs, with ROUTE set to N.

Reference:

"Insert Call Processing SM (RC/V 5.81)" (5-31)

5 At the MCC or TLWS type and enter the backup command.

```
BKUP:ODD;
```

Result:

```
BKUP ODD   COMPLETED
```

6 Update RC/V 5.81 to change the ROUTE flags from N to Y. (This two-step update allows the SM time to initialize dynamic data before calls start routing to the SM).

Reference:

"Insert Call Processing SM (RC/V 5.81)" (5-31)

7 At the MCC or TLWS type and enter the backup command.

```
BKUP:ODD;
```

Result:

```
BKUP ODD   COMPLETED
```

8 Stop. You have completed this procedure.

END OF STEPS

□

Add a SIP PHE2 to Existing Simplex SIP IP Processor Group

Purpose

This section contains the procedure for adding a SIP PHE2 to an existing simplex SIP IP processor group.

When to use

Use this procedure to add a SIP PHE2 to an existing simplex SIP IP processor group.

Related information

Refer to 235-118-258, Recent Change Reference, for information on the RC/V system.

Refer to 235-080-100, Translations Guide, for information on Office Records (output reports).

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Admonishments

Conducting an MML “switch”command in this procedure has the potential to lose transient calls.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release with the SIP for Packet Trunking - NAR feature software.

Required Information

- STBY PHE2 Hardware (SM, PSU Shelf, and Position)
- Processor Group (SM, PSU, and PCRGRP)

Required Conditions

- Verify that the hardware growth procedures have been executed for the desired PHE2 board, and that it is in the STBY state on MCC page 118x, where ″x″ is the shelf on which the PHE2 board resides.

Procedure

1 Select and prepare terminal for RC/V activities.

Reference:

“Select and Prepare Terminal” (5-11)

2 On RC/V 22.16 for the PSU shelf where the STBY PHE2 exists, select a channel group number, update the PH TYPE to PHE2, and update the RMK field.

Reference:

“PH Channel Group Assignment (RC/V 22.16)” (5-19)

3 Review RC/V 33.16 to obtain the other PH in the Processor Group.

4 Read RC/V 33.1 for the existing channel group in the processor group, and execute a change/insert operation to change *QUALIFIER 3 and the local IP address.

5 Examine the ETHIP/5995 office records and select an unused PHE2 Ethernet Link number.

6 Update RC/V 33.16 with the selected PHE2 shelf, position, channel group, and Ethernet link number.

This operation automatically inserts RC/V 33.4 with the new channel group, Ethernet IP interface name, and parameters identical to the existing simplex PHE2 channel group in the processor group.

Shortly after this update, there will be a REPT PSELNK output indicating that the Ethernet link has been established. (If this does not happen, debug the Ethernet and router connections until it does.)

If the previously-existing PH in the processor group was not SERVING (or UNAVAILABLE because either its PHE2 or Ethernet link were OOS), there will also be an autonomous REPT SERV ROP output indicating that the new PH in the processor group is SERVING. Otherwise, the new PH should become NON-SERVING.

7 At the MCC or TLWS type and enter the backup command.

```
BKUP:ODD;
```

Result:

```
BKUP ODD   COMPLETED
```

8 At the MCC or TLWS, poke or enter the “switch” command SW:PSUPH= to ensure that the PHs are working as expected.

9 Stop. You have completed this procedure.

END OF STEPS

Change IP/SCTP Transport Address for SCTP Near Endpoint

Purpose

This section contains the procedure for changing the IP/SCTP transport address for an SCTP near endpoint.

When to use

Use this procedure to change the IP/SCTP transport address for an SCTP near endpoint.

Related information

Refer to 235-118-258, Recent Change Reference, for information on the RC/V system.

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release with the SIP for Packet Trunking - NAR feature software.

Required Information

- SCTP Near Endpoint name
- New IP address(es)

Required Conditions

- Use OP:STATUS,SCTP,NEAREPT to verify that the near endpoint is in the OOS MAN state. If not, use RMV:SCTP, NEAREPT to put it in the OOS MAN state.

Procedure

1 Select and prepare terminal for RC/V activities.

Reference:

“Select and Prepare Terminal” (5-11)

2 Type and enter the RC/V form number.

`33.19`

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

3 Type and enter the review command.

`R`

Result:

SCTP NEAR ENDPOINT DEFINITION page is displayed and the cursor is located at NEAR ENDPOINT NAME field.

4 Using the RC/V 33.19 form as a guide, type and enter the indicated value.

- *NEAR ENDPOINT NAME

5 Record the processor group.

6 Type and enter the previous screen command.

`<`

Result:

INTERNET PROTOCOL VIEWS page is displayed.

7 Type and enter the RC/V form number.

`33.16`

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

8 Type and enter the review command.

`R`

Result:

PROCESSOR GROUP page is displayed and the cursor is located at PCR GRP field

9 Using the RC/V 33.16 form as a guide, type and enter the indicated values for each field.

- *PCR GRP
- *SM - SIP GSM number

10 Record the channel group(s).

11 Type and enter the previous screen command.

<

Result:

INTERNET PROTOCOL VIEWS page is displayed.

12 Type and enter the RC/V form number.

`33.4`

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

13 Type and enter the update command.

`U`

Result:

Screen 1 of the ETHERNET INTERNET PROTOCOL (IP) INTERFACE ASSIGNMENT page is displayed and the cursor is located at the SM field.

14 Using screen 1 of the RC/V 33.4 form as a guide and the data from Step 13, type and enter the parameters.

- *SM - Switching Module number
- *PSU - Packet Switching Unit number
- *SHELF - PSU shelf number
- *CHANNEL GROUP - Channel group position

15 Proceed to screen 2.

```
2
```

Result:

Screen 2 is displayed and the cursor is located at the GATEWAY IP ADDR1 field.

---

16 On screen 2 of RC/V 33.4, update the IP addresses assigned to the Ethernet IP interface.

---

17 Type and enter the update command.

```
U
```

Result:

```
Updating...form updated
```

---

18 Type and enter the previous screen command.

```
<
```

Result:

The INTERNET PROTOCOL VIEWS page is displayed.

---

19 If desired, type and enter the quit command to exit the RC/V system.

```
Q
```

Result:

The RC/V session is terminated.

---

20 At the MCC or TLWS type and enter the backup command.

```
BKUP:ODD;
```

Result:

```
BKUP ODD   COMPLETED
```

---

21 Verify that all far offices with associations to the updated near endpoint have made equivalent updates to the IP addresses (the SCTP near endpoint of an association on the *5ESS*® switch is the SCTP far endpoint of the equivalent association in the far office).

22 When both ends have their ODDs in synch as to the IP addresses of the SCTP endpoint, type and enter:

`RST:SCTP,NEAREPT;`

Result:

Establishes all the associations that were not individually CLOSED MAN prior to this procedure.

Then type and enter: `OP:STATUS,SCTP,NEAREPT,DETAIL;` to verify that the associations were established.

At this point, any stable calls that survived may be able to signal to the far end again.

23 Stop. You have completed this procedure.

END OF STEPS

Change IP/SCTP Transport Address for SCTP Far Endpoint

Purpose

This section contains the procedure for changing the IP/SCTP transport address for an SCTP far endpoint.

When to use

Use this procedure to change the IP/SCTP transport address for an SCTP far endpoint.

Related information

Refer to 235-118-258, Recent Change Reference, for information on the RC/V system.

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release with the SIP for Packet Trunking - NAR feature software.

Required Information

- Far Endpoint name
- New IP address(es)

Required Conditions

- Use OP:STATUS,SCTP,ASSOC,ALL to verify that the SCTP associations to the affected far endpoint (from all SCTP near endpoints, on all SIP-T GSMs) are in the CLOSED MAN state. If they are not, use the procedure to put them in the CLOSED MAN state.

Procedure

1 Select and prepare terminal for RC/V activities.

Reference:

“Select and Prepare Terminal” (5-11)

2 Type and enter the RC/V form number.

```
33.21
```

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

3 Type and enter the update command.

```
U
```

Result:

SCTP FAR ENDPOINT DEFINITION page is displayed and the cursor is located at FAR ENDPOINT NAME field.

4 Using the RC/V 33.21 form as a guide, type and enter the indicated values for each field. (Updates may require multiple steps. Enter "C" for change and then enter field number to update each desired non-key field.)

- *FAR ENDPOINT NAME
- #DEST SCTP PORT
- #DEST IP ADDR 1
- DEST IP ADDR 2 - (optional)

5 Enter the update command.

```
U
```

Result:

```
Updating...form updated
```

6 Type and enter the previous screen command.

```
<
```

Result:

INTERNET PROTOCOL VIEWS page is displayed.

7 At the MCC or TLWS type and enter the backup command.

```
BKUP:ODD;
```

Result:

```
BKUP ODD   COMPLETED
```

8 Verify that the far office has completed the same IP address update for its SCTP near endpoint.

9 Use the RST:SCTP,ASSOC command to restore the associations to the far endpoint.

10 Use the OP:STATUS,SCTP,ASSOC command to verify that they have been established.

11 Stop. You have completed this procedure.

END OF STEPS

Change Far IP Transport Address for UDP Path

Purpose

This section contains the procedure for changing the IP address and/or UDP port of the far end of a UDP path.

When to use

Use this procedure when the IP transport address or the UDP port number of the far end office changes, and a UDP path exists to the old address.

Related information

Refer to 235-118-258, Recent Change Reference, for information on the RC/V system.

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS®* switch must be running a 5E16.2 FR5 software release with UDP Transport for SIP software.

Required Information

- New IP address and UDP port for SIP signaling at far office
- UDP Path number of existing UDP Path to far office

Required Conditions

- None

Procedure

1 Examine the PKTGRP/5219 records to determine whether there is a SIP Packet Group assigned to the UDP path in question. If there is no SIP Packet Group assigned to the UDP Path, proceed to step 6.

2 If a SIP Packet Group is assigned to the UDP path, examine the TKGRP/5202 office records to determine whether there is a trunk group assigned to the SIP Packet Group. If no trunk group is assigned, proceed to step 5.

3 If a Trunk Group is assigned to the SIP Packet Group, examine the RTIDX/5303 office records to determine whether any route indices point to the trunk group. Record the information from all such forms, for later use. Update or delete the ROUTE INDEX (ROUTING) RC/V 10.2 for any such route indices until there are no more route indices pointing to the trunk group.

4 Record the information for the trunk group for later re-insertion, and delete the Trunk Group on RC/V 5.1.

5 Record the information for the SIP packet group for later re-insertion, and delete the SIP Packet Group on RC/V 5.71.

6 If the new far IP address is not in the same IP subnet as the IP address of the Ethernet-IP interface of the PHs in the processor group for the UDP path, use `OP:TCPIP,RTDMP` and/or examine the IPROUT/5989 office records to verify that the necessary IP route is provisioned, or execute the “Insert IP Routing (RC/V 33.3)” procedure.

7 Use `EXC:PING` to verify the reachability of the new far IP address from the near IP interface.

8 Update the FAR IP ADDR and/or FAR PORT on the RC/V 33.24 UDP PATH view.

9 If updates/deletes were made on RC/V 10.2 ROUTE INDEX, RC/V 5.1 TRUNK GROUP, and/or RC/V 5.71 PACKET GROUP DEFINITION, use the recorded information to update and/or re-insert those views as required. Refer to the “Add New SIP Packet Trunking to Another Office” procedure.

10 Execute `BKUP:ODD`

11 Stop. You have completed this procedure.

END OF STEPS

Change IP Address for Adjacent IP Router

Purpose

This section contains the procedure for changing the IP address for an adjacent IP router.

When to use

Use this procedure to change the IP address for an adjacent IP router.

Related information

Refer to 235-118-258, Recent Change Reference, for information on the RC/V system.

Refer to 235-080-100, Translations Guide, for information on Office Records (output reports).

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking - NAR feature software.

Required Information

- Old and New router IP addresses

Required Conditions

- None

Procedure

1 Dump the IPPING/5884 office records. At the MCC or TLWS type and enter the command:

`OP:OFR:FORM=5884`

Result:

OP:OFR:FORM=5884 PF is printed followed by a printout of the IPPING/5884 office records.

2 Identify the IP interface(s) names in the IPPING/5884 office records that have ping data for the old address of the adjacent IP router.

3 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

4 Type and enter the RC/V form number.

```
33.17
```

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

5 Type and enter the delete command.

```
D
```

Result:

The INTERNET PROTOCOL (IP) PING PARAMETERS page is displayed and the cursor is located at the INTERFACE NAME field.

6 Using the RC/V 33.17 form as a guide, type and enter the parameters.

- *INTERFACE NAME - name identifying interface
- *IP ADDRESS - address of the access router

7 Enter the delete command.

```
D
```

Result:

```
Deleting...form deleted
```

8 Dump the IPROUT/5989 office records. At the MCC or TLWS type and enter the command:

```
OP:OFR:FORM=5989
```

Result:

OP:OFR:FORM=5989 PF is printed followed by a printout of the IPROUT/5989 office records.

9 Identify all the combinations of Ethernet IP interface and DEST IP ADDR that route via the old IP address of the adjacent IP router in the “GATEWAY IP ADDRESS” field.

10 Type and enter the RC/V form number.

`33.3`

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

11 Type and enter the update command.

`U`

Result:

The INTERNET PROTOCOL (IP) ROUTING TO INTERFACE page is displayed and the cursor is located at the DEST IP ADDR field

12 Using the RC/V 33.3 form as a guide, type and enter the parameters. (Updates may require multiple steps. Enter “C” for change and then enter field number to update each desired non-key field.)

- *DEST IP ADDR - destination IP address that can be reached with this route
- *INTERFACE NAME - name of defined interface
- #GATEWAY IP ADDR - IP address of the gateway through which data is sent to the destination

13 Enter the update command.

`U`

Result:

```
Updating...form Updated
```

14 Repeat steps 11-13 for each IP address identified in step 9

15 Type and enter the previous screen command.

<

Result:

The INTERNET PROTOCOL VIEWS page is displayed.

16 Insert the IP ping data for the new IP address from each of the Ethernet IP interface names previously identified.

Reference:

"Insert Router Pinging (RC/V 33.17)" (5-47)

17 At the MCC or TLWS type and enter the backup command.

`BKUP:ODD;`

Result:

```
BKUP ODD   COMPLETED
```

18 Stop. You have completed this procedure.

END OF STEPS

Change IP Parameters for a SIP processor group

Purpose

This section contains the procedure for changing the SIP processor group parameters.

When to use

Use this procedure to change the IP parameters for a SIP processor group.

Related information

Refer to 235-118-258, Recent Change Reference, for information on the RC/V system.

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking - NAR feature software.

Required Information

- SM, PSU, and PCRGRP number

Required Conditions

- None

Procedure

1 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

2 Type and enter the RC/V form number.

`33.16`

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

3 Type and enter the review command.

`R`

Result:

PROCESSOR GROUP page is displayed and the cursor is located at PCR GRP field

4 Using the RC/V 33.16 form as a guide, type and enter the indicated values for each field.

- *PCR GRP
- *SM - SIP GSM number

5 Identify and record the channel group(s) in the processor group.

6 Type and enter the previous screen command.

`<`

Result:

INTERNET PROTOCOL VIEWS page is displayed.

7 Are Processor IP parameter changes required?

- If yes, continue with the next step.
- If no, proceed to step 19 to skip Processor IP parameter changes.

8 Type and enter the RC/V form number.

`33.1`

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

9 Type and enter the update command.

`U`

Result:

Screen 1 of the INTERNET PROTOCOL (IP) PROCESSOR ASSIGNMENT page is displayed and the cursor is located at PROCESSOR ID field.

10 Using screen 1 of RC/V 33.1 form as a guide, type and enter the indicated values for each field.

- *PROCESSOR ID - SM number
- *PROCESSOR TYPE - PH
- *QUALIFIER 2 - SM number
- *QUALIFIER 3 - 1 digit shelf number and 2 digit channel group number concatenated.

11 Type and enter screen 2.

```
2
```

Result:

Screen 2 of the INTERNET PROTOCOL (IP) PROCESSOR ASSIGNMENT page is displayed

12 Using screen 2 of RC/V 33.1 modify the fields as desired.

13 Proceed to screen 3.

```
3
```

Result:

Screen 3 of the INTERNET PROTOCOL (IP) PROCESSOR ASSIGNMENT page is displayed and the cursor is located at the ICMP ERR GEN field

14 Using screen 3 of RC/V 33.1 modify the fields as desired.

15 Enter the update command.

```
U
```

Result:

```
Warnings may be displayed depending on the changes
   made.
```

16 Enter the ignore command.

```
I
```

Result:

```
Ignoring...form Updated.
```

17 Type and enter the previous page command.

```
<
```

Result:

INTERNET PROTOCOL VIEWS page is displayed.

18 Are Ethernet Interface IP parameter changes required?

- If yes, continue with the next step.
- If no, proceed to step 26 to skip the Ethernet interface IP parameter changes.

19 Type and enter the RC/V form number.

```
33.4
```

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

20 Type and enter the update command.

```
U
```

Result:

Screen 1 of the ETHERNET INTERNET PROTOCOL (IP) INTERFACE ASSIGNMENT page is displayed and the cursor is located at the SM field.

21 Using screen 1 of the RC/V 33.4 form as a guide, type and enter the parameters.

- *SM - Switching Module number
- *PSU - Packet Switching Unit number
- *SHELF - PSU shelf number
- *CHANNEL GROUP - Channel group position

22 Proceed to screen 2.

`2`

Result:

Screen 2 is displayed.

23 On screen 2 of RC/V 33.4, modify the parameters as desired.

24 Type and enter the update command.

`U`

Result:

`Updating...form updated`

25 Type and enter the previous screen command.

`<`

Result:

The INTERNET PROTOCOL VIEWS page is displayed.

26 If desired, type and enter the quit command to exit the RC/V system.

`Q`

Result:

The RC/V session is terminated.

27 At the MCC or TLWS type and enter the backup command.

`BKUP:ODD;`

Result:

```
BKUP ODD   COMPLETED
```

28 Stop. You have completed this procedure.

END OF STEPS

Change SCTP Endpoint Parameters

Purpose SCTP Endpoint Parameters (RC/V 33.18) view is used to update the SCTP endpoint related timers and protocol parameters.

When to use Use this procedure to update the SCTP endpoint parameters.

Related information Refer to 235-118-258, Recent Change Reference, for information on the RC/V system.

Admonishments None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking - NAR feature software.

Required Information

- SCTP Endpoint parameter set name

Required Conditions

- None

Procedure

1 Select and prepare terminal for RC/V activities.

Reference:

“Select and Prepare Terminal” (5-11)

2 Type and enter the RC/V form number.

```
33.18
```

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

3 Type and enter the update command.

`U`

Result:

The SCTP ENDPOINT PARAMETERS page is displayed and the cursor is located at the PARM SET NAME field.

4 Using the RC/V 33.18 form as a guide, type and enter.

- *PARM SET NAME

5 Modify the parameters as needed.

6 Enter the update command.

`U`

Result:

`Updating...form updated`

7 Type and enter the previous screen command.

<

Result:

The INTERNET PROTOCOL VIEWS page is displayed.

8 If desired, type and enter the quit command to exit the RC/V system.

`Q`

Result:

The RC/V session is terminated.

9 At the MCC or TLWS type and enter the backup command.

`BKUP:ODD;`

Result:

`BKUP ODD   COMPLETED`

10 Stop. You have completed this procedure.

END OF STEPS

Change SCTP Association Parameters

Purpose

This section contains the procedure for changing the SCTP association parameters.

When to use

Use this procedure to change the SCTP association parameters.

Related information

Refer to 235-118-258, Recent Change Reference, for information on the RC/V system.

Refer to 235-080-100, Translations Guide, for information on Office Records (output reports).

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking - NAR feature software.

Required Information

- SCTP Association parameter set name

Required Conditions

- None

Procedure

1 All associations that use the parameter set to be updated, must be CLOSED MAN before the update can be made.

Examine the SCTPASSC/5889 office records to find all the associations that reference the association parameter set to be updated, and use RMV:SCTP,ASSOC to put them all in the CLOSED MAN state. The procedure can be used to remove ***all*** SCTP Associations or in this case to delete ***selected*** Associations.

---

2 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

---

3 Type and enter the RC/V form number.

`33.20`

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

---

4 Type and enter the update command.

`U`

Result:

The SCTP ASSOCIATION PARAMETER SET page is displayed.

---

5 Using the RC/V 33.20 form as a guide, type and enter the parameter set name in the *PARM SET NAME field.

---

6 Modify the parameters as needed.

---

7 Enter the update command.

`U`

Result:

```
Updating...form updated
```

---

8 Type and enter the previous screen command.

`<`

Result:

The INTERNET PROTOCOL VIEWS page is displayed.

9 If desired, type and enter the quit command to exit the RC/V system.

```
Q
```

Result:

The RC/V session is terminated.

10 If associations were CLOSED MAN in order to update the parameters, use RST:SCTP,ASSOC to establish them again.

11 At the MCC or TLWS type and enter the backup command.

```
BKUP:ODD;
```

Result:

```
BKUP ODD   COMPLETED
```

12 Stop. You have completed this procedure.

END OF STEPS

Change SIP Parameters

Purpose

The SIP-T Parameters Definition view (RC/V 5.82) is used to change the SIP parameter definitions.

When to use

Use this procedure to change a SIP parameter set.

Related information

Refer to 235-118-258, Recent Change Reference, for information on the RC/V system.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking - NAR feature software.

Required Information

- SIP Parameter set name

Required Conditions

- If this RC/V view is being updated to support **SIP without Encapsulated ISUP**, the *5E-XC*™ must be running on software release 5E16.2 FR6 or later. In addition, SFID 769 must be activated prior to executing this procedure. To activate a secured feature refer to "Feature Activation (RC/V 8.22)" (5-13).

Procedure

1 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

2 Type and enter the RC/V form number.

```
5.82
```

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

3 Type and enter the update command.

```
U
```

Result:

The SIP-T PARAMETERS DEFINITION page is displayed.

4 Type and enter.

- *PARM SET NAME

5 Using the RC/V 5.82 form as a guide, modify the timers and parameters as required.

NOTE: The service provider is responsible for ensuring proper coordination of its switches. If SIP without Encapsulated ISUP is being provisioned, the packet group at the TPS which will receive the INVITE without ISUP must have TRUST ISUP set to **no**. Trust ISUP=N is the proper setting when connecting a TPS to a proxy and there are multiple network elements on the other side with varying capabilities. Trust ISUP=Y is the proper setting when two ‘true’ switches that understand ISUP are directly connected.

NOTE: The service provider is responsible for ensuring proper coordination of its switches. If SIP without Encapsulated ISUP is being provisioned, the packet group at the TPS which will receive the INVITE without ISUP must have an appropriate SIP to PSTN Interworking Parameter set (RC/V View 5.83) assigned. Refer to the *Feature Description*, 235-190-400, document for SIP to PSTN Interworking Parameter set guidelines.

6 Enter the update command.

```
U
```

Result:

```
Updating...form updated
```

7 Type and enter the previous screen command.

<

Result:

The TRUNK VIEWS page is displayed.

8 If desired, type and enter the quit command to exit the RC/V system.

`Q`

Result:

The RC/V session is terminated.

9 At the MCC or TLWS type and enter the backup command.

`BKUP:ODD;`

Result:

```
BKUP ODD   COMPLETED
```

10 Stop. You have completed this procedure.

END OF STEPS

Provision Alarm Level for IP Fragmented Packets Beyond PM Threshold

Purpose

The IP Fragmented Packets alarm alerts the switch personnel of a potential security problem. The Threshold Alarm Level view (RC/V 8.29) is used to enable or disable the alarm.

When to use

This procedure is executed to enable the IP Fragmented Packets Alarm. By default the alarm is disabled.

Related information

Refer to 235-118-258, Recent Change Reference for information on the RC/V system.

Admonishments

Disabling the alarm could cause a security problem to go unnoticed by the switch personnel.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking - NAR feature software.

Required Information

- None.

Required Conditions

- None.

Procedure

1 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

2 Type and enter the RC/V form number.

```
8.29
```

Result:

```
Enter Database Operation R=Review,U=Update:
```

3 Type and enter the update command.

```
U
```

Result:

The THRESHOLD ALARM LEVELS page is displayed.

4 Using the RC/V 8.29 form as a guide, type and enter the parameters.

- KEY - Any value
- IP FRAG DATAGRM - MINOR

5 Type and enter the update command.

```
U
```

Result:

Updating...form updated

6 Type and enter the previous screen command

```
<
```

Result:

The OFFICE MISC. & ALARMS VIEWS page is displayed.

7 Type and enter the RC/V form number.

```
20.32
```

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

8 Type and enter the insert command.

```
I
```

Result:

Screen 1 of the OIU PERFORMANCE MONITORING THRESHOLD GROUP page is displayed.

9 Using the RC/V 20.32 form as a guide, type and enter the parameters.

- PM GROUP - performance monitoring group
- CKT TYPE - Type of OIU facility the PM group can have assigned to it.

Result:

Screen 2 of the RC/V 20.32 is displayed.

10 Using screen 2 of RC/V 20.32 as a guide, type and enter the parameters.

- IIF15MAT - Number of incoming fragmented IP datagrams (1-900)

11 Type and enter the insert command.

```
I
```

Result:

```
Inserting...form inserted
```

12 Type and enter the previous screen command.

```
<
```

Result:

SM PACK & SUBPACK VIEWS page is displayed.

13 If desired, type and enter the quit command to exit the RC/V system.

```
Q
```

Result:

The RC/V session is terminated.

14 Stop. You have completed this procedure.

END OF STEPS

Provision Alarm Level for ICMP Echo Requests Beyond PM Threshold

Purpose

ICMP echo request alarm alerts the switch personnel of a potential security problem. The Threshold Alarm Level view (RC/V 8.29) is used to enable or disable the alarm.

When to use

This procedure is executed to enable the ICMP Echo Request Alarm. By default the alarm is disabled.

Related information

Refer to 235-118-258, Recent Change Reference for information on the RC/V system.

Admonishments

Disabling the alarm could cause a security problem to go unnoticed by the switch personnel.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking - NAR feature software.

Required Information

- None

Required Conditions

- Secured Feature 684 is enabled (RC/V 8.22)
- SIP GSM is defined (RC/V 5.80)

Procedure

1 Select and prepare terminal for RC/V activities.

Reference:

“Select and Prepare Terminal” (5-11)

2 Type and enter the RC/V form number.

`8.29`

Result:

```
Enter Database Operation,R=Review,U=Update:
```

3 Type and enter the update command.

`U`

Result:

The THRESHOLD ALARM LEVELS page is displayed.

4 Using the RC/V 8.29 form as a guide, type and enter the parameters.

- KEY - Any value
- ICMP ECHO REQ - MINOR

5 Type and enter the update command.

`U`

Result:

```
Updating...form updated
```

6 Type and enter the previous screen command

`<`

Result:

The OFFICE MISC. & ALARMS VIEW is displayed.

7 Type and enter the RC/V form number.

`20.32`

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

8 Type and enter the insert command.

`I`

Result:

Screen 1 of the OIU PERFORMANCE MONITORING THRESHOLD GROUP page is displayed.

---

9 Using the RC/V 20.32 form as a guide, type and enter the parameters.

- PM GROUP - Performance monitoring group name
- CKT TYPE - Type of OIU facility the PM group can have assigned to it.

Result:

Screen 2 of the OIU PERFORMANCE MONITORING THRESHOLD GROUP page is displayed.

---

10 Using screen 2 of RC/V 20.32 as a guide, type and enter the parameters.

- ICEDAT - Number of ICMP echo requests received before alarm triggers (1-65535)

---

11 Type and enter the insert command.

`I`

Result:

`Inserting...form inserted`

---

12 Type and enter the previous screen command.

`<`

Result:

SM PACK AND SUBPACK VIEWS menu page is displayed.

---

13 If desired, type and enter the quit command to exit the RC/V system.

`Q`

Result:

The RC/V session is terminated.

---

14 Stop. You have completed this procedure.

END OF STEPS

Change SIP PSTN Interworking Parameter Set

Purpose The SIP PSTN Interworking Parameter Set view (RC/V 5.83) is used to change the SIP PSTN interworking parameter definitions.

When to use Use this procedure to change a SIP PSTN interworking parameter set.

Related information Refer to 235-118-258, Recent Change Reference, for information on the RC/V system.

Admonishments None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR6 software release or later with the SIP for Packet Trunking - NAR feature software.

Required Information

- SIP PSTN Interworking Parameter set name

Required Conditions

- If this RC/V view is being updated to support **SIP without Encapsulated ISUP**, the *5E-XC*™ must be running on software release 5E16.2 FR6 or later. In addition, SFID 769 must be activated prior to executing this procedure. To activate a secured feature refer to “Feature Activation (RC/V 8.22)” (5-13).

Procedure

1 Select and prepare terminal for RC/V activities.

Reference:

“Select and Prepare Terminal” (5-11)

2 Type and enter the RC/V form number.

```
5.83
```

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

3 Type and enter the update command.

```
U
```

Result:

The SIP PSTN INTERWORKING PARAMETER SET page is displayed.

4 Type and enter.

- *SET NAME

5 Using the RC/V 5.83 form as a guide, modify the timers and parameters as required.

6 Enter the update command.

```
U
```

Result:

```
Updating...form updated
```

7 Type and enter the previous screen command.

```
<
```

Result:

The TRUNK VIEWS page is displayed.

8 If desired, type and enter the quit command to exit the RC/V system.

```
Q
```

Result:

The RC/V session is terminated.

9 At the MCC or TLWS type and enter the backup command.

```
BKUP:ODD;
```

Result:

```
BKUP ODD   COMPLETED
```

10 Stop. You have completed this procedure.

END OF STEPS

6 Deprovisioning

Overview

Purpose

This chapter contains the procedures for deprovisioning the Session Initiation Protocol application in the *5ESS*® switch.

The *5ESS*® switch Recent Change and Verify (RC/V) interface provides access to the database through views of the Office Dependent Data (ODD). Only those views and steps necessary to deprovision SIP are covered in this chapter. For complete descriptions of RC/V views or more information about recent change and verify, refer to 235-118-258, Recent Change Reference, and 235-118-251, Recent Change Procedures.

The Deprovisioning section does not include the procedures for deprovisioning traffic measurements. This can be found in 235-070-100, Administration and Engineering Guidelines.

Additionally, this section does not include procedures to degrow hardware. Refer to 235-105-331, Hardware Change Procedures - Degrowth for information on degrowth of the PHE2 and PH33 hardware.

All deprovisioning procedures described in this section are performed using RC/V menu mode.

The RC/V menu mode can be entered from the RC/V terminal or a Supplemental Trunk Line Workstation (STLWS).

At the end of this chapter there are additional Deprovisioning procedures that may be required when deleting and/or changing certain aspects of your already existing SIP signaling network. Please reference the “Delete SIP Signaling from an Existing SIP Network” (6-88) procedures.

Deprovisioning Sequence

Introduction

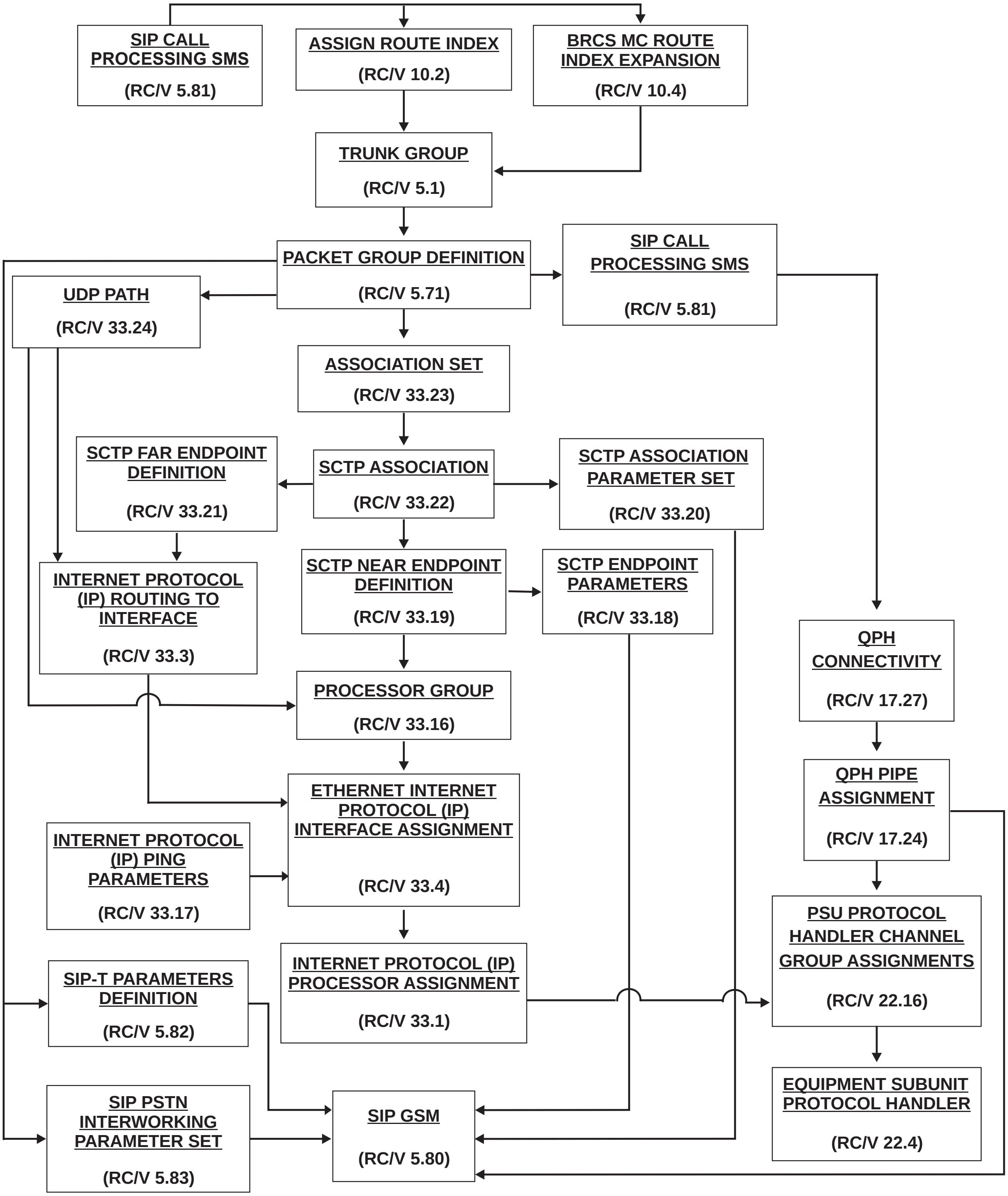
This section contains the procedures for deprovisioning the SM connectivity and signaling. Figure 6-1, “Deprovisioning Flowchart” (6-4) contains a flowchart that indicates the order the RC/V views are executed and the “Deprovisioning Sequence” (6-5) contains steps with more details for deprovisioning the SIP application.

Note:The SM connectivity deprovisioning does not apply on DRM/VCDX.

Refer to 235-105-331, Hardware Change Procedures - Degrowth for information on OIU-IP degrowth procedures to delete IP bearer, if IP packet trunking is no longer to be used.

Deprovisioning Flowchart The following flowchart shows the deprovisioning sequence.

Figure 6-1 Deprovisioning Flowchart



Deprovisioning Sequence

Below is a sequence of steps for deprovisioning the SIP application. This section assumes the reader is knowledgeable about the RC/V system and the reading of office records. More procedural information can be found in the links that are listed within the steps.

1. "Disable INVITE Requests (RC/V 5.81)" (6-8)
   On RC/V 5.81, set the "ROUTE" to "N" to prevent any new SIP calls from being originated or terminated.
2. "Disable All SCTP Transport for SIP Packet Trunking" (6-11).
3. "Remove All SIP Trunk Groups From Route Lists (RC/V 10.2)" (6-14)
   On RC/V 10.2, remove all SIP trunk groups from all route lists identified.
4. "Remove All SIP Trunk Groups From Route Lists (RC/V 10.4)" (6-17)
   On RC/V 10.4, remove all SIP trunk groups from all route lists identified.
5. "Delete All SIP Packet Trunk Groups (RC/V 5.1)" (6-20)
   On RC/V 5.1, delete all trunk groups that have the SIPT GRP field set to Y.
6. "Delete All SIP Packet Groups (RC/V 5.71)" (6-23)
   On RC/V 5.71, delete all SIP packet groups.
7. "Disable All UDP Transport for SIP Packet Trunking" (6-26)
8. "Delete All SIP Call Processing SMs (RC/V 5.81)" (6-29)
   Delete each instance of RC/V 5.81 identified previously.
9. " Delete All SCTP Association Sets (RC/V 33.23)" (6-32)
   On RC/V 33.23, delete all identified SCTP association sets.
10. " Delete All SCTP Associations (RC/V 33.22)" (6-35)
    On RC/V 33.22, delete all SCTP associations.
11. "Delete All SCTP Far Endpoints (RC/V 33.21)" (6-38)
    On RC/V 33.21, delete all identified SCTP far endpoints.
12. "Delete All SCTP Near Endpoints (RC/V 33.19)" (6-41)
    On RC/V 33.19, delete all identified SCTP near endpoints.
13. "Delete All SCTP Association Protocol Parameter Sets (RC/V 33.20)" (6-44)
    On RC/V 33.20, delete all identified SCTP association parameter sets.

14. “Delete All SCTP Endpoint Timers and Protocol Parameters (RC/V 33.18)” (6-47)
    On RC/V 33.18, delete all SCTP endpoint parameter sets identified.
15. “Delete All Router Pinging (RC/V 33.17)” (6-50)
    On RC/V 33.17, delete router pinging parameters.
16. “Delete All SIP UDP Paths (RC/V 33.24)” (6-53)
    Delete All SIP UDP Paths (RC/V 33.24)
17. “Delete All IP Routing for SIP UDP Paths (RC/V 33.3)” (6-55)
    For each IPROUT/5989 record, if the IP interface matches one of the IP interface names identified, delete the corresponding IP ROUTING TO INTERFACE instance on RC/V 33.3.
18. “Delete All IP Routing for SCTP Associations (RC/V 33.3)” (6-59)
    For each IPROUT/5989 record, if the IP interface matches one of the IP interface names identified, delete the corresponding IP ROUTING TO INTERFACE instance on RC/V 33.3.
19. “Delete All SIP Processor Groups (RC/V 33.16)” (6-63)
    On RC/V 33.16, delete all of the SIP processor groups identified previously.
20. “Delete IP Processor Assignment to PH (RC/V 33.1)” (6-66) On RC/V 33.1 delete IP addresses and parameters to a processor.
21. “Update All SIP GSM - Non-GSM Communication (RC/V 17.27)” (6-69)
    For each QGCON/5829 record where the SERVICE is SIPT, update RC/V 17.27 to remove all the Non-Global SM-2000s from the list, except for the one that matches the GSM, because that has to stay as long as there are GQPH pipes provisioned.
22. “ Delete All SIP GQPH Pipes (RC/V 17.24)” (6-72)
    For each QPHPIPE in the QPHPIPE/5828 office record where SERVICE is SIPT, delete the QPHPIPE on RC/V 17.24.
23. “Update All PHE2 & PH33 Channel Group Assignments (RC/V 22.16)” (6-75)
    On RC/V 22.16, for each PSU shelf identified, for each channel group with PH TYPE set to PH33 and GRP TYPE set to NULL, update the PH TYPE TO NULL.
24. “Delete All SIP Parameter Sets (RC/V 5.82)” (6-78)
    On RC/V 5.82, delete all SIP parameter sets except for the DEFAULT parameter set.

25. "Delete All SIP PSTN Interworking Parameter Sets (RC/V 5.83)" (6-81)
    On RC/V 5.83, delete all SIP PSTN interworking parameter sets except for the DEFAULT parameter set.
26. "Delete All SIP Global SMs (RC/V 5.80)" (6-84)
    On RC/V 5.80, delete the SIP GSM. This should automatically delete the DEFAULT SIP parameter set. Check RC/V 5.82 and delete it manually if it exists.
27. "Delete Protocol Handler (RC/V 22.4)" (6-87)

☐

Disable INVITE Requests (RC/V 5.81)

Purpose

Disabling ″INVITE″ requests should prevent any SIP calls from being originated or terminated. Outgoing calls will not be routed on SIP packet groups. The SIP-T Call Processing SMs view (RC/V 5.81) defines SM-2000s that can serve as a call processing SM for SIP calls.

When to use

Use this procedure to disable an SM-2000 from processing SIP calls. Execute this step for each SIP GSM.

Related information

Refer to 235-118-258, Recent Change Reference for information on the RC/V system.

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking - NAR feature software.

Required Information

- GSM number(s)

Required Conditions

- None

Procedure

1 Use the OP:OFR command to print the SIPTSMS/5881 office records. Identify the GSM numbers that have SM-2000(s) with the ROUTE field set to Y.

At the MCC or TLWS type and enter the command:

```
OP:OFR:FORM=5881;
```

Result:

```
The OP OFR FORM=5881 is printed followed by a
   printout of the SIPTSMS/5581 records.
```

---

2 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

---

3 Type and enter the RC/V form number.

```
5.81
```

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

---

4 Type and enter the update command.

```
U
```

Result:

The SIP-T CALL PROCESSING SMS page is displayed and the cursor is located at the GSM field.

---

5 Using the RC/V 5.81 form as a guide, type and enter the GSM number.

Result:

```
The SIP-T SMS (SIPTSMS) table lists the SM-2000
   MODULES and ROUTE status for the SM-2000s
   associated with the GSM
```

---

6 Using the RC/V 5.81 form as a guide, change the ROUTE field to N for all SM-2000 modules. (This should prevent any new SIP calls from being originated or terminated. Incoming ″INVITE″ requests for SIP calls will be denied, on a call-by-call basis, and outgoing calls will not be routed on SIP packet groups. Execute this step for each SIP GSM.)

- ROUTE - N

7 Enter the updated command.

`U`

Result:

`Updating...form updated`

NOTE: If GSM number equals SM2K number, a Warning message is displayed.

`“SM2K Module in SIP-T SMS list should not equal GSM. This is not applicable for VCDX/DRM office.”`

If GSM should/does equal non-GSM enter “I” to ignore the warning and continue.

8 Repeat steps 5-7 for each GSM identified in step 1.

9 Type and enter the previous screen command.

`<`

Result:

The TRUNKS VIEWS page is displayed.

10 If desired, type and enter the quit command to exit the RC/V system.

`Q`

Result:

The RC/V session is terminated.

11 At the MCC or TLWS type and enter the backup command.

`BKUP:ODD;`

Result:

`BKUP ODD   COMPLETED`

12 Stop. You have completed this procedure.

END OF STEPS

Disable All SCTP Transport for SIP Packet Trunking

Purpose

This procedure prevents new incoming or outgoing SIP calls from being established over the SCTP transport layer.

With all associations removed from service (in the CLOSED MAN state), far offices should mark all SIP packet groups to this office as out-of-service, and incoming SIP call attempts should cease.

When to use

This procedure should only be used to prevent any incoming or outgoing calls from using SIP packet trunking with SCTP transport, generally only if the whole SIP packet trunking feature is being degrown from the office.

Related information

Refer to 235-118-258, Recent Change Reference for information on the RC/V system.

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Admonishments

- If there is no equivalent circuit route (such as an ISUP7 trunk group) to a particular far office that is served by SIP packet trunking, this procedure will cause it to be impossible to route calls to that far office.
- If an SM Full Initialization of a SIP GSM occurs, the effects of this procedure are automatically reversed by the initialization, and this procedure will need to be repeated.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking - NAR feature software.

Required Information

- Association numbers of all provisioned SCTP associations. These can be obtained either from examining the office records (FORM=5889), or from the first step of the procedure.

Required Conditions

- None

Procedure

1 Execute the following command to verify the status of all provisioned associations:

```
OP:STATUS,SCTP,ASSOC,ALL;
```

Result:

The “OP STATUS SCTP ASSOC ALL” output report is printed, consisting of multiple segments of status information for all provisioned SCTP associations.

2 For each provisioned association number in the output of the previous step, that is not already in the CLOSED MAN state, execute the following command.

```
RMV:SCTP,ASSOC=xxxx,UCL;
```

where,

xxxx = the association number

Result:

The “RMV SCTP ASSOC=XXXX” output report is printed, indicating that the association is in the CLOSED MAN state.

The “REPT SCTP ENDPOINTT=yyyyyy autonomous output report may be printed if the removal of the association caused a status change for its SCTP near endpoint (from INSERV to PARTSERV when the first association on the endpoint is removed, and from PARTSERV to NOSERV when the last association on the endpoint is removed).

3 Repeat steps 1 and 2 until the result from step 1 indicates that all provisioned SCTP associations are in the CLOSED MAN state.

Result:

SCTP transport for SIP packet trunking is no longer operational.

4 Stop. You have completed this procedure.

END OF STEPS

Remove All SIP Trunk Groups From Route Lists (RC/V 10.2)

Purpose

The Route Index view (RC/V 10.2) is used to delete route indices. A route index determines a network path, typically used to complete a call or route to error treatment when the call is unable to complete.

When to use

Use this procedure to remove all route indices that point to trunk groups used for SIP-T packet groups.

Related information

Refer to 235-118-258, Recent Change Reference for information on the RC/V system.

Refer to 235-080-100, Translations Guide, for information on Office Records (output reports).

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Admonishments

Subscribers can no longer access SIP trunk groups.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking - NAR feature software.

Required Information

- All Route indices assigned to SIP-T packet trunk groups.

Required Conditions

- None.

Procedure

1 Identify the TRUNK GROUP numbers that have SIPT GRP set to Y. At the MCC or TLWS type and enter the command:
`OP:OFR:FORM=5894;`

Result:

```
OP OFR FORM=5894 PF is printed followed by a printout
    of the INTGI/5894 office records.
```

2 Dump the RTIDX/5303 office records. At the MCC or TLWS type and enter the command:

```
OP:OFR:FORM=5303;
```

Result:

```
The OP OFR FORM=5303 is printed followed by a
   printout of the RTINDX/5303 records.
```

3 Identify the route indices associated with the trunk groups identified in Step 1.

4 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

5 Type and enter the RC/V form number.

```
10.2
```

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

6 Type and enter the delete command.

```
D
```

Result:

Screen 1 of the ROUTE INDEX (ROUTING) page is displayed and the cursor is at the RTI field.

7 Using the RC/V 10.2 form as a guide, type and enter the parameters identified in step 2.

- *RTI - route index (1-16382)

8 Type and enter the delete command.

`D`

Result:

`Deleting...form deleted`

9 Repeat steps 7 and 8 for every route and trunk group identified in steps 1 and 2.

10 Type and enter the previous screen command.

`<`

Result:

ROUTING & CHARGING VIEWS page is displayed.

11 If desired, type and enter the quit command to exit the RC/V system.

`Q`

Result:

The RC/V session is terminated.

12 At the MCC or TLWS type and enter the backup command.

`BKUP:ODD;`

Result:

`BKUP ODD   COMPLETED`

13 Stop. You have completed this procedure.

END OF STEPS

Remove All SIP Trunk Groups From Route Lists (RC/V 10.4)

Purpose

The BRCS MC Route Index Expansion view (RC/V 10.4) is used to delete MC Route Indices. The MC Route Index may be defined for accessing a trunk group which is used as a private facility, an SFG or as overflow to POTS routing.

When to use

Use this procedure to delete all MC route indices that point to trunk groups used by SIP-T packet groups.

Related information

Refer to 235-118-258, Recent Change Reference for information on the RC/V system.

Refer to 235-080-100, Translations Guide, for information on Office Records (output reports).

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking - NAR feature software.

Required Information

- All MC Route Index numbers assigned to SIP-T packet trunk groups.

Required Conditions

- None.

Procedure

1 Identify the TRUNK GROUP numbers for trunk groups that have SIPT GRP set to Y. At the MCC or TLWS type and enter the command:

```
OP:OFR:FORM=5894;
```

Result:

```
OP OFR FORM=5894 PF is printed followed by a printout
    of the INTGI/5894 office records.
```

2 Dump the MCRTIDX/5304 office records. At the MCC or TLWS type and enter the command:

```
OP:OFR:FORM=5304;
```

Result:

```
OP OFR FORM=5304 PF is printed followed by a printout
    of the MCRTIDX/5304 office records.
```

3 Identify the route indices associated with the trunk groups identified in Step 1.

4 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

5 Type and enter the RC/V form number.

```
10.4
```

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

6 Type and enter the delete command.

```
D
```

Result:

Screen 1 of the BRCS MC ROUTE INDEX EXPANSION page is displayed and the cursor is at the MC RTIDX field.

7 Using the RC/V 10.4 form as a guide, type and enter the parameters identified in step 2.

- *MC RTIDX - route index (1-16382)

8 Type and enter the delete command.

`D`

Result:

`Deleting...form deleted`

9 Repeat steps 7 and 8 for every route and trunk group identified in steps 1 and 2.

10 Type and enter the previous screen command.

`<`

Result:

ROUTING & CHARGING VIEWS page is displayed.

11 If desired, type and enter the quit command to exit the RC/V system.

`Q`

Result:

The RC/V session is terminated.

12 At the MCC or TLWS type and enter the backup command.

`BKUP:ODD;`

Result:

`BKUP ODD   COMPLETED`

13 Stop. You have completed this procedure.

END OF STEPS

Delete All SIP Packet Trunk Groups (RC/V 5.1)

Purpose The Trunk Group view (RC/V 5.1) is used to delete trunk groups.

When to use Use this procedure to delete all trunk groups used for SIP-T packet trunking.

Related information Refer to 235-118-258, Recent Change Reference for information on the RC/V system.

Refer to 235-080-100, Translations Guide, for information on Office Records (output reports).

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Admonishments None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking - NAR feature software.

Required Information

- All Trunk Group numbers used for SIP-T packet trunking.

Required Conditions

- All Route indices that refer to SIP-T packet trunk groups on 10.2 and 10.4 have been deleted.

Procedure

1 Identify the TRUNK GROUP numbers for trunk groups that have SIPT GRP set to Y. At the MCC or TLWS type and enter the command:

```
OP:OFR:FORM=5894;
```

Result:

```
OP OFR FORM=5894 PF is printed followed by a printout
    of the TKGRP/5894 office records.
```

2 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

3 Type and enter the RC/V form number.

`5.1`

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

4 Type and enter the delete command.

`D`

Result:

Screen 1 of the TRUNK GROUP page is displayed.

5 Type in the TRUNK GROUP number and enter the delete command.

`D`

Result:

```
Deleting...form deleted
```

6 Repeat step 5 for every trunk group that has the SIPT GRP field set to Y identified in step 1.

7 Type and enter the previous screen command.

`<`

Result:

TRUNKS VIEWS page is displayed.

8 If desired, type and enter the quit command to exit the RC/V system.

`Q`

Result:

The RC/V session is terminated.

9 At the MCC or TLWS type and enter the backup command.

`BKUP:ODD;`

Result:

```
BKUP ODD   COMPLETED
```

10 Stop. You have completed this procedure.

END OF STEPS

Delete All SIP Packet Groups (RC/V 5.71)

Purpose

The Packet Group view (RC/V 5.71) is used to delete a SIP packet group. A SIP packet group is used to designate packet trunking to a given office.

When to use

Use this procedure to delete all SIP packet groups.

Related information

Refer to 235-118-258, Recent Change Reference for information on the RC/V system.

Refer to 235-080-100, Translations Guide, for information on Office Records (output reports).

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking - NAR feature software.

Required Information

- All SIP-T Packet Group numbers

Required Conditions

- All trunk groups that refer to SIP-T packet groups have been deleted on RC/V 5.1.

Procedure

1 Identify the PKT GRP numbers for trunk groups that have SIPT GRP set to Y. At the MCC or TLWS type and enter the command:
`OP:OFR:FORM=5219;`

Result:

```
OP OFR FORM=5219 PF is printed followed by a printout
    of the PKTGRP/5219 office records.
```

---

2 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

---

3 Type and enter the RC/V form number.

```
5.71
```

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

---

4 Type and enter the delete command.

```
D
```

Result:

The PACKET GROUP DEFINITION page is displayed and the cursor is located at the PKT GRP field.

---

5 Using the RC/V 5.71 form as a guide, type and enter a PKT GRP number identified in step 1.

---

6 Enter the delete command.

```
D
```

Result:

```
Deleting...form deleted
```

---

7 Repeat steps 5-6 for every packet group identified in steps 1 and 2.

---

8 Type and enter the previous screen command.

```
<
```

Result:

The TRUNK VIEWS page is displayed.

9 If desired, type and enter the quit command to exit the RC/V system.

`Q`

Result:

The RC/V session is terminated.

10 At the MCC or TLWS type and enter the backup command.

`BKUP:ODD;`

Result:

```
BKUP ODD   COMPLETED
```

11 Stop. You have completed this procedure.

END OF STEPS

□

Disable All UDP Transport for SIP Packet Trunking

Purpose

This procedure prevents new incoming or outgoing SIP calls from being established over the UDP transport layer.

Since UDP paths are not configurable apart from the processor group to which they belong, it is necessary to remove the PHs in the processor groups in order to make the IP address of the processor group unreachable from the IP network. Removing the SIP PHs from service will cause the routers in the IP network to report “Destination Unreachable” to any switch that attempts to send SIP messages to them. That same router report will let the other offices know that they should avoid attempting new calls via SIP/UDP to the unavailable PHs.

When to use

This procedure should only be used to prevent any incoming or outgoing calls from using SIP packet trunking with UDP transport, generally only if the whole SIP packet trunking feature is being degrown from the office.

Related information

Refer to 235-118-258, Recent Change Reference for information on the RC/V system.

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Admonishments

- If there is no equivalent circuit route (such as an ISUP7 trunk group) to a particular far office that is served by SIP packet trunking, this procedure will cause it to be impossible to route calls to that far office.
- If an SM Full Initialization of a SIP GSM occurs, the effects of this procedure are automatically reversed by the initialization, and this procedure will need to be repeated.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking - NAR feature software.

Required Information

- Identification of all provisioned UDP paths. These can be obtained either from examining the office records (FORM=5891), or from the first step of the procedure.

Required Conditions

- None

Procedure

1 Execute the following command to dump the UDPPATH office records:

`OP:OFR:FORM=5891;`

Result:

`OP OFR FORM=5219 PF is displayed and an output report is printed, consisting of multiple segments of status information for all provisioned UDP paths.`

2 Record the SM and PROCESSOR GROUP for each UDP path.

3 For each PROCESSOR GROUP identified in the previous step, execute the following command:

`OP:STATUS,SERV,PCRGRP=xxx-yy;`

where,

xxx = the SM number, and

yy = the processor group number.

4 Use the `RMV:PSUPH` command, with the UCL option, for each PSUPH identified in the OP STATUS SERV output from the previous step. First remove all the NONSERVING PHs, then the SERVING PHs, until all processor groups are UNAVAILABLE.

Result:

UDP transport for SIP packet trunking is no longer operational.

5 Stop. You have completed this procedure.

END OF STEPS

Delete All SIP Call Processing SMs (RC/V 5.81)

Purpose

The SIP-T Call Processing SMs view (RC/V 5.81) defines SM-2000s that can serve as a call processing SM for SIP calls.

When to use

Use this procedure to delete all SM-2000 SIP-T call processing SMs.

Related information

Refer to 235-118-258, Recent Change Reference for information on the RC/V system.

Refer to 235-080-100, Translations Guide, for information on Office Records (output reports).

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Refer to 235-105-331, Hardware Change Procedures - Degrowth for information on degrowth of the PHE2.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking - NAR feature software.

Required Information

- All SIP-T GSM numbers

Required Conditions

- SM-2000 module numbers disabled in the Disable INVITE Requests (RC/V 5.81)
- All SIP-T packet groups deleted on 5.71

Procedure

1 Identify all SIP GSMs that have SIP CP SMs associated with them. At the MCC or TLWS type and enter the command:

```
OP:OFR:FORM=5881;
```

Result:

```
OP:OFR:FORM=5881  PF is printed followed by a
   printout of the 5881 office records.
```

2 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

3 Type and enter the RC/V form number.

```
5.81
```

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

4 Type and enter the delete command.

```
D
```

Result:

The SIP-T CALL PROCESSING SMS page is displayed and the cursor is located at the GSM field.

5 Using the RC/V 5.81 form as a guide, type and enter the parameters.

- * GSM

6 Enter the delete command.

```
D
```

Result:

```
Deleting...form deleted
```

7 Repeat steps 5 and 6 for every SM-2000 SIP-T GSM identified in Step 1.

8 Type and enter the previous screen command.

```
<
```

Result:

The TRUNKS VIEWS page is displayed.

---

9 If desired, type and enter the quit command to exit the RC/V system.

`Q`

Result:

The RC/V session is terminated.

---

10 At the MCC or TLWS type and enter the backup command.

`BKUP:ODD;`

Result:

```
BKUP ODD   COMPLETED
```

---

11 Stop. You have completed this procedure.

END OF STEPS

Delete All SCTP Association Sets (RC/V 33.23)

Purpose

SCTP Association Set (RC/V 33.23) groups associations into association set for redundancy and load-sharing purposes.

When to use

Use this procedure to delete all association sets.

Related information

Refer to 235-118-258, Recent Change Reference for information on the RC/V system.

Refer to 235-080-100, Translations Guide, for information on Office Records (output reports).

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking - NAR feature software.

Required Information

- All Association set names

Required Conditions

- All Packet Groups that refer to SCTP association sets have been deleted on RC/V 5.71

Procedure

1 Dump the ASSCSET/5890 office records. At the MCC or TLWS type and enter the command:

```
OP:OFR:FORM=5890;
```

Result:

```
OP OFR FORM=5890 PF is printed followed by a printout
    the ASSCSET/5890 office records.
```

2 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

3 Type and enter the RC/V form number.

`33.23`

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

4 Type and enter the delete command.

`D`

Result:

The ASSOCIATION SET page is displayed and the cursor is located at ASSOCIATION SET NAME field.

5 Using the RC/V 33.23 form as a guide, type and enter the values for each field.

- *ASSOCIATION SET NAME

6 Enter the insert command.

`D`

Result:

```
Deleting...form deleted
```

7 Repeat steps 5 and 6 for every association set identified in step 1.

8 Type and enter the previous screen command.

`<`

Result:

The INTERNET PROTOCOL VIEWS page is displayed.

9 If desired, type and enter the quit command to exit the RC/V system.

```
Q
```

Result:

The RC/V session is terminated.

---

10 At the MCC or TLWS type and enter the backup command.

```
BKUP:ODD;
```

Result:

```
BKUP ODD   COMPLETED
```

---

11 Stop. You have completed this procedure.

END OF STEPS

Delete All SCTP Associations (RC/V 33.22)

Purpose SCTP Association (RC/V 33.22) defines an association between near and far SCTP endpoints.

When to use Use this procedure to delete all SCTP associations.

Related information Refer to 235-118-258, Recent Change Reference for information on the RC/V system.

Refer to 235-080-100, Translations Guide, for information on Office Records (output reports).

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Admonishments None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking - NAR feature software.

Required Information

- All SCTP Association numbers

Required Conditions

- All Association sets have been deleted on RC/V 33.23

Procedure

1 Dump the SCTPASSC/5889 office records. At the MCC or TLWS type and enter the command:

```
OP:OFR:FORM=5889;
```

Result:

```
OP OFR FORM=5889 PF is printed followed by a printout
    the SCTPASSC/5889 office records.
```

2 Select and prepare terminal for RC/V activities.

Reference:

“Select and Prepare Terminal” (5-11)

3 Type and enter the RC/V form number.

`33.22`

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

4 Type and enter the delete command.

`D`

Result:

SCTP ASSOCIATION page is displayed and the cursor is located at ASSOC NUMBER field

5 Using the RC/V 33.22 form as a guide, type and enter the values for each field.

- *ASSOC NUMBER

6 Enter the delete command.

`D`

Result:

`Deleting...form deleted`

7 Repeat steps 5 and 6 for every association identified in step 1.

8 Type and enter the previous screen command.

`<`

Result:

INTERNET PROTOCOL VIEWS page is displayed.

9 If desired, type and enter the quit command to exit the RC/V system.

```
Q
```

Result:

The RC/V session is terminated.

10 At the MCC or TLWS type and enter the backup command.

```
BKUP:ODD;
```

Result:

```
BKUP ODD   COMPLETED
```

11 Stop. You have completed this procedure.

END OF STEPS

Delete All SCTP Far Endpoints (RC/V 33.21)

Purpose

SCTP Far Endpoint (RC/V 33.21) view is used to delete SCTP far endpoints.

When to use

Use this procedure to delete all of the far SCTP endpoints.

Related information

Refer to 235-118-258, Recent Change Reference for information on the RC/V system.

Refer to 235-080-100, Translations Guide, for information on Office Records (output reports).

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking - NAR feature software.

Required Information

- All SCTP Far endpoint names

Required Conditions

- All SCTP associations have been deleted on RC/V 33.22

Procedure

1 Dump the SCTPFEPD/5888 office records. At the MCC or STLWS type and enter the command:

```
OP:OFR:FORM=5888
```

Result:

```
OP OFR FORM=5888 PF is printed followed by a printout
    of the SCTPFEPD/5888 office records.
```

2 Select and prepare terminal for RC/V activities.

Reference:

“Select and Prepare Terminal” (5-11)

3 Type and enter the RC/V form number.

`33.21`

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

4 Type and enter the delete command.

`D`

Result:

SCTP FAR ENDPOINT DEFINITION page is displayed and the cursor is located at FAR ENDPOINT NAME field.

5 Using the RC/V 33.21 form as a guide, type and enter the *FAR ENDPOINT NAME.

6 Enter the delete command.

`D`

Result:

```
Deleting...form deleted
```

7 Repeat steps 5-6 for every far endpoint identified in step 1.

8 Type and enter the previous screen command.

`<`

Result:

INTERNET PROTOCOL VIEWS page is displayed.

9 If desired, type and enter the quit command to exit the RC/V system.

`Q`

Result:

The RC/V session is terminated.

10 At the MCC or TLWS type and enter the backup command.

```
BKUP:ODD;
```

Result:

```
BKUP ODD   COMPLETED
```

11 Stop. You have completed this procedure.

END OF STEPS

Delete All SCTP Near Endpoints (RC/V 33.19)

Purpose The SCTP Endpoint (RC/V 33.19) view is used to delete the SCTP near endpoints to a processor group.

When to use Use this procedure to delete the near SCTP endpoints.

Related information Refer to 235-118-258, Recent Change Reference for information on the RC/V system.

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Admonishments None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking - NAR feature software.

Required Information

- All SCTP Near Endpoint names

Required Conditions

- All SCTP associations have been deleted on RC/V 33.22

Procedure

1 Identify the SCTP near endpoints to delete.

Enter the command:

`OP:STATUS,SCTP,NEAREPT,ALL;`

Result:

```
The OP STATUS SCTP NEARPT report is printed. The
   status of each near endpoint is GROW, indicating
   that the required condition that all SCTP
   associations have been deleted on RC/V 33.22 has
   been met.
```

2 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

3 Type and enter the RC/V form number.

`33.19`

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

4 Type and enter the delete command.

`D`

Result:

SCTP NEAR ENDPOINT DEFINITION page is displayed and the cursor is located at NEAR ENDPOINT NAME field.

5 Using the RC/V 33.19 form as a guide, type and enter the indicated values for each field.

- *NEAR ENDPOINT NAME

6 Enter the delete command.

`D`

Result:

```
Deleting...form deleted
```

7 Repeat steps 5 and 6 for each near endpoint identified in step 1.

8 Type and enter the previous screen command.

`<`

Result:

INTERNET PROTOCOL VIEWS page is displayed.

9 If desired, type and enter the quit command to exit the RC/V system.

```
Q
```

Result:

The RC/V session is terminated.

10 At the MCC or TLWS type and enter the backup command.

```
BKUP:ODD;
```

Result:

```
BKUP ODD   COMPLETED
```

11 Stop. You have completed this procedure.

END OF STEPS

Delete All SCTP Association Protocol Parameter Sets (RC/V 33.20)

Purpose

The SCTP Association Parameter Set (RC/V 33.20) view is used to update protocol parameters for an SCTP association.

When to use

This procedure should be followed to delete customer defined association parameter sets.

Related information

Refer to 235-118-258, Recent Change Reference for information on the RC/V system.

Refer to 235-080-100, Translations Guide, for information on Office Records (output reports).

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking - NAR feature software.

Required Information

- All SCTP Association Parameter set names.

Required Conditions

- All SCTP Associations have been deleted on RC/V 33.22

Procedure

1 Dump the ASSCPARM/5887 office records. At the MCC or TLWS type and enter the command:

```
OP:OFR:FORM=5887;
```

Result:

```
OP OFR FORM=5887 PF is printed followed by a printout
    of the ASSCPARM/5887 office records.
```

2 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

3 Type and enter the RC/V form number.

```
33.20
```

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

4 Type and enter the delete command.

```
D
```

Result:

The SCTP ASSOCIATION PARAMETER SET page is displayed and the cursor is located at the PARM SET NAME field.

5 Using the RC/V 33.20 form as a guide, type and enter the parameter set name in the *PARM SET NAME field.

6 Enter the delete command.

```
D
```

Result:

```
Deleting...form Deleted
```

7 Repeat steps 5 and 6 for every SCTP association parameter set.

8 Type and enter the previous screen command.

```
<
```

Result:

The INTERNET PROTOCOL VIEWS page is displayed.

9 If desired, type and enter the quit command to exit the RC/V system.

`Q`

Result:

The RC/V session is terminated.

10 At the MCC or TLWS type and enter the backup command.

`BKUP:ODD;`

Result:

```
BKUP ODD   COMPLETED
```

11 Stop. You have completed this procedure.

END OF STEPS

Delete All SCTP Endpoint Timers and Protocol Parameters (RC/V 33.18)

Purpose

SCTP Endpoint Parameters (RC/V 33.18) is used to delete the SCTP endpoint related timers and protocol parameters.

When to use

This procedure is used to delete all SCTP Endpoint parameter sets.

Related information

Refer to 235-118-258, Recent Change Reference for information on the RC/V system.

Refer to 235-080-100, Translations Guide, for information on Office Records (output reports).

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking - NAR feature software.

Required Information

- All SCTP Endpoint Parameter Set Names.

Required Conditions

- All SCTP Near Endpoints have been deleted on RC/V 33.19

Procedure

1 Dump the SCTPPARM/5885 office records. At the MCC or TLWS type and enter the command:

```
OP:OFR:FORM=5885;
```

Result:

```
OP OFR FORM=5885 PF is printed followed by a printout
    of the SCTPPARM/5885 office records.
```

2 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

3 Type and enter the RC/V form number.

```
33.18
```

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

4 Type and enter the delete command.

```
D
```

Result:

The SCTP ENDPOINT PARAMETERS page is displayed and the cursor is located at the PARM SET NAME field.

5 Using the RC/V 33.18 form as a guide, type and enter the *PARM SET NAME.

Important! The DEFAULT SCTP parameter set must be deleted last.

6 Enter the delete command.

```
D
```

Result:

```
Deleting...form deleted
```

7 Repeat step 5 and 6 for every SCTP endpoint parameter set identified in step 1.

8 Type and enter the previous screen command.

```
<
```

Result:

The INTERNET PROTOCOL VIEWS page is displayed.

9 If desired, type and enter the quit command to exit the RC/V system.

```
Q
```

Result:

The RC/V session is terminated.

10 At the MCC or TLWS type and enter the backup command.

```
BKUP:ODD;
```

Result:

```
BKUP ODD   COMPLETED
```

11 Stop. You have completed this procedure.

END OF STEPS

Delete All Router Pinging (RC/V 33.17)

Purpose

The Internet Protocol (IP) Ping Parameters view (RC/V 33.17) is used to delete pinging parameters pertaining to a router through a particular interface.

When to use

Use this procedure to delete the pinging parameters of all access routers via Ethernet IP interfaces on SIP PHs.

Related information

Refer to 235-118-258, Recent Change Reference for information on the RC/V system.

Refer to 235-080-100, Translations Guide, for information on Office Records (output reports).

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking - NAR feature software.

Required Information

- IP Address of the access router

Required Conditions

- None.

Procedure

1 Dump the PCRGRP/5883 office records. At the MCC or TLWS type and enter the command:

```
OP:OFR:FORM=5883;
```

Result:

```
OP OFR FORM=5883 PF is printed followed by a printout
   of the PCRGRP/5883 office records.
```

2 Dump the ETHIP/5995 office records. At the MCC or TLWS type and enter the command:

```
OP:OFR:FORM=5995;
```

Result:

```
The ETHIP/5995 office records are printed.
```

3 Identify the IP INTERFACE NAME for each ETHIP/5995 office record where the SM, PSU, SHELF and CHGRP matches the SM, PSU, SHELF and CHGRP in the PCRGRP/5883 office record.

4 Dump the IPPING/5884 office records. At the MCC or TLWS type and enter the command:

```
OP:OFR:FORM=5884
```

Result:

```
OP:OFR:FORM=5884 PF is printed followed by a printout
   of the IPPING/5884 office records.
```

5 Identify the IP interface(s) in the IPPING/5884 office record that match one of the IP interface names of the SIP PHE2(s) in step 3, and each router IP address associated with them.

6 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

7 Type and enter the RC/V form number.

```
33.17
```

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
  D=Delete:
```

8 Type and enter the delete command.

D

Result:

The INTERNET PROTOCOL (IP) PING PARAMETERS page is displayed and the cursor is located at the INTERFACE NAME field.

---

9 Using the RC/V 33.17 form as a guide, type and enter the parameters.

- *INTERFACE NAME - name identified in step 5
- *IP ADDRESS - address of the access router identified in step 5

---

10 Enter the delete command.

D

Result:

```
Deleting...form deleted
```

---

11 Repeat steps 9-10 for each IP interface/IP address pair identified in step 5.

---

12 Type and enter the previous screen command.

<

Result:

The INTERNET PROTOCOL VIEWS page is displayed.

---

13 If desired, type and enter the quit command to exit the RC/V system.

Q

Result:

The RC/V session is terminated.

---

14 At the MCC or TLWS type and enter the backup command.

BKUP:ODD;

Result:

```
BKUP ODD   COMPLETED
```

---

15 Stop. You have completed this procedure.

END OF STEPS

Delete All SIP UDP Paths (RC/V 33.24)

Purpose

The UDPPATH view (RC/V 33.24) provides provisioning information for all UDP paths.

When to use

Use this procedure to delete UDP paths.

Related information

Refer to 235-118-258, Recent Change Reference for information on the RC/V system.

Refer to 235-080-100, Translations Guide, for information on Office Records (output reports).

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking - NAR feature software.

Required Information

- None

Required Conditions

- All SIP Packet groups have been deleted on RC/V 5.71

Procedure

1 Dump the UDPPATH office records with all the SIP UDP paths. At the MCC or TLWS type and enter the command:

```
OP:OFR:FORM=5891;
```

Result:

```
OP OFR FORM=5891 PF is printed followed by a printout
    of the UDPPATH/5891 office records.
```

2 Record the GSM & PROCESSOR GROUP for each UDP PATH, to be referred to in the “Delete all IP routing for SIP UDP Paths” procedure which follows.

3 Use RC/V 33.24 to delete each SIP UDP path identified in the office records from FORM 5891 recorded above.

4 Backup the office records. At the MCC or TLWS type and enter the command:

`BKUP:ODD;`

5 Stop. You have completed this procedure.

END OF STEPS

Delete All IP Routing for SIP UDP Paths (RC/V 33.3)

Purpose

The Internet Protocol (IP) Routing view (RC/V 33.3) provides the capability to deprovision an IP gateway between an external IP destination and a local IP interface.

When to use

Use this procedure to delete IP gateways between external IP destinations and local IP interfaces.

Related information

Refer to 235-118-258, Recent Change Reference for information on the RC/V system.

Refer to 235-080-100, Translations Guide, for information on Office Records (output reports).

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking - NAR feature software.

Required Information

- Destination IP address.
- Gateway IP address

Required Conditions

- All SIP UDP paths have been deleted on RC/V 33.24

Procedure

1 Dump the PCRGRP forms for the processor groups recorded in the “Delete All SIP UDP Paths” procedure. At the MCC or TLWS type and enter the command:

```
OP:OFR:FORM=5883;
```

Result:

```
OP OFR FORM=5883 PF is printed followed by a printout
    of the PCRGRP/5883 office records.
```

---

2 Dump the ETHIP/5995 office records for each SM-PSU-SHELF-CHGRP combination from the PCRGRP forms identified above, and record the INTERFACE NAMEs. At the MCC or TLWS type and enter the command:

```
OP:OFR:FORM=5995;
```

Result:

```
OP OFR FORM=5995 PF is printed followed by a printout
    of the ETHIP/5995 office records.
```

---

3 Identify the IP INTERFACE NAME for each ETHIP/5995 office record where the SM, PSU, SHELF and CHGRP matches the SM, PSU, SHELF and CHGRP in the PCRGRP/5883 office record.

---

4 Dump the IPROUT/5989 office records. At the MCC or TLWS type and enter the command:

```
OP:OFR:FORM=5989;
```

Result:

```
OP:OFR:FORM=5989 PF is printed followed by a printout
    of the IPROUT/5989 office records.
```

---

5 Identify the IP interface(s) in the IPROUT/5989 office record that match one of the IP interface names of the SIP PHE2(s) in step 3, and each destination IP address associated with them.

---

6 Select and prepare terminal for RC/V activities.

Reference:

“Select and Prepare Terminal” (5-11)

---

7 Type and enter the RC/V form number.

```
33.3
```

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

8 Type and enter the delete command.

```
D
```

Result:

The INTERNET PROTOCOL (IP) ROUTING TO INTERFACE page is displayed and the cursor is located at the DEST IP ADDR field.

9 Using the RC/V 33.3 form as a guide, type and enter the parameters.

- *DEST IP ADDR - destination IP address that can be reached with this route (identified in step 5)
- *INTERFACE NAME - name of defined interface identified in step 5

10 Enter the delete command.

```
D
```

Result:

```
Deleting...form deleted
```

11 Repeat steps 9-10 for every IP interface/destination IP address pair identified in step 5.

12 Type and enter the previous screen command.

```
<
```

Result:

The INTERNET PROTOCOL VIEWS page is displayed.

13 If desired, type and enter the quit command to exit the RC/V system.

```
Q
```

Result:

The RC/V session is terminated.

14 At the MCC or TLWS type and enter the backup command.

```
BKUP:ODD;
```

Result:

```
BKUP ODD   COMPLETED
```

15 Stop. You have completed this procedure.

END OF STEPS

Delete All IP Routing for SCTP Associations (RC/V 33.3)

Purpose

The Internet Protocol (IP) Routing view (RC/V 33.3) provides the capability to deprovision an IP gateway between an external IP destination and a local IP interface.

When to use

Use this procedure to delete IP gateways between external IP destinations and local IP interfaces.

Related information

Refer to 235-118-258, Recent Change Reference for information on the RC/V system.

Refer to 235-080-100, Translations Guide, for information on Office Records (output reports).

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking - NAR feature software.

Required Information

- Destination IP address.
- Gateway IP address

Required Conditions

- All SCTP Far Endpoints have been deleted on RC/V 33.21

Procedure

1 Dump the PCRGRP/5883 office records. At the MCC or TLWS type and enter the command:

`OP:OFR:FORM=5883;`

Result:

```
OP OFR FORM=5883 PF is printed followed by a printout
    of the PCRGRP/5883 office records.
```

2 Dump the ETHIP/5995 office records. At the MCC or TLWS type and enter the command:

```
OP:OFR:FORM=5995;
```

Result:

```
OP OFR FORM=5995 PF is printed followed by a printout
    of the ETHIP/5995 office records.
```

3 Identify the IP INTERFACE NAME for each ETHIP/5995 office record where the SM, PSU, SHELF and CHGRP matches the SM, PSU, SHELF and CHGRP in the PCRGRP/5883 office record.

4 Dump the IPROUT/5989 office records. At the MCC or TLWS type and enter the command:

```
OP:OFR:FORM=5989;
```

Result:

```
OP:OFR:FORM=5989 PF is printed followed by a printout
    of the IPROUT/5989 office records.
```

5 Identify the IP interface(s) in the IPROUT/5989 office record that match one of the IP interface names of the SIP PHE2(s) in step 3, and each destination IP address associated with them.

6 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

7 Type and enter the RC/V form number.

```
33.3
```

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

8 Type and enter the delete command.

D

Result:

The INTERNET PROTOCOL (IP) ROUTING TO INTERFACE page is displayed and the cursor is located at the DEST IP ADDR field.

9 Using the RC/V 33.3 form as a guide, type and enter the parameters.

- *DEST IP ADDR - destination IP address that can be reached with this route (identified in step 5)
- *INTERFACE NAME - name of defined interface identified in step 5

10 Enter the delete command.

D

Result:

```
Deleting...form deleted
```

11 Repeat steps 9-10 for every IP interface/destination IP address pair identified in step 5.

12 Type and enter the previous screen command.

<

Result:

The INTERNET PROTOCOL VIEWS page is displayed.

13 If desired, type and enter the quit command to exit the RC/V system.

Q

Result:

The RC/V session is terminated.

14 At the MCC or TLWS type and enter the backup command.

BKUP:ODD;

Result:

```
BKUP ODD   COMPLETED
```

15 Stop. You have completed this procedure.

END OF STEPS

Delete All SIP Processor Groups (RC/V 33.16)

Purpose

The Processor Group view (RC/V 33.16) view is used to delete all SIP processor groups.

When to use

Use this procedure when SIP is being degrown from the office entirely.

Related information

Refer to 235-118-258, Recent Change Reference for information on the RC/V system.

Refer to 235-080-100, Translations Guide, for information on Office Records (output reports).

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Admonishments

The PH's will revert to STBY mode on MCC page 118X,Y,X.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking - NAR feature software.

Required Information

- All Processor Group Numbers on all SIP GSMs

Required Conditions

- All SCTP Near Endpoints deleted on RC/V 33.19
- All UDP paths deleted on RC/V 33.24

Procedure

1 Dump the PCRGRP/5883 office records. At the MCC or TLWS type and enter the command:

```
OP:OFR:FORM=5883;
```

Result:

```
OP OFR FORM=5883 PF is printed followed by a printout
    of the PCRGRP/5883 office records.
```

2 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

3 Type and enter the RC/V form number.

```
33.16
```

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

4 Type and enter the delete command.

```
D
```

Result:

PROCESSOR GROUP page is displayed and the cursor is located at PCR GRP field

5 Using the RC/V 33.16 form as a guide, type and enter the indicated values for each field identified in step 1.

- *PCR GRP
- *SM - SIP GSM number

6 Enter the delete command.

```
D
```

Result:

```
Deleting...form deleted
```

7 Repeat steps 5-6 for every processor group identified in step 1.

8 Type and enter the previous screen command.

```
<
```

Result:

INTERNET PROTOCOL VIEWS page is displayed.

9 If desired, type and enter the quit command to exit the RC/V system.

```
Q
```

Result:

The RC/V session is terminated.

10 At the MCC or TLWS type and enter the backup command.

```
BKUP:ODD;
```

Result:

```
BKUP ODD   COMPLETED
```

11 Stop. You have completed this procedure.

END OF STEPS

Delete IP Processor Assignment to PH (RC/V 33.1)

Purpose

The Internet Protocol (IP) Processor Assignment view (RC/V 33.1) is used to delete IP processor assignments and parameters on a processor (PHE2).

When to use

Use this procedure to delete IP parameters on a processor.

Repeat individual steps of this procedure as required.

Related information

Refer to 235-118-258, Recent Change Reference, for information on the RC/V system.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking - NAR feature software.

Required Information

- Processor ID
- Processor type
- Shelf number
- Channel Group number

Required Conditions

- PROCESSOR GROUP (PCRGRP/5883/33.16 form) has been deleted

Procedure

1 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

2 Type and enter the RC/V form number.

```
33.1
```

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

3 Type and enter the delete command.

```
D
```

Result:

Screen 1 of the INTERNET PROTOCOL (IP) PROCESSOR ASSIGNMENT page is displayed and the cursor is located at PROCESSOR ID field.

4 Using screen 1 of RC/V 33.1 form as a guide, delete each field.

- *PROCESSOR ID - SM number
- *PROCESSOR TYPE - SM (or PH)
- *QUALIFIER 2 - SM number (or for a PH, this field is the PSU community address (COM ADDR) found on RC/V 22.2, Packet Switch Unit).
- *QUALIFIER 3 - for an SM leave blank (or for a PH, 1 digit shelf number and 2 digit channel group number concatenated).

5 Type and enter the delete command.

```
D
```

Result:

```
Deleting...form deleted
```

6 Repeat steps 4 and 5 for each IP processor assignment.

7 Type and enter the previous page command.

```
<
```

Result:

INTERNET PROTOCOL VIEWS page is displayed.

8 If desired, type and enter the quit command to exit the RC/V system.

`Q`

Result:

The RC/V system is The RC/V session is terminated.

9 At the MCC or TLWS type and enter the backup command.

`BKUP:ODD;`

Result:

```
BKUP ODD   COMPLETED
```

10 Stop. You have completed this procedure.

END OF STEPS

Update All SIP GSM - Non-GSM Communication (RC/V 17.27)

Purpose

The General GSM/NGSM Connectivity view (RC/V 17.27) defines the logical links between the GSM and Non-GSM.

When to use

Use this procedure to remove connectivity to Non-Global SM-2000s with the SIP service type from all GSMs.

NOTE: This is not needed for DRM/VCDX environment.

Related information

Refer to 235-118-258, Recent Change Reference for information on the RC/V system.

Refer to 235-080-100, Translations Guide, for information on Office Records (output reports).

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking - NAR feature software.

Required Information

- All GSM numbers that have SIP-T service connectivity to NGSMs.

Required Conditions

- All SIP-T SMs have been deleted on RC/V 5.81

Procedure

1 Identify the QGCON/5829 office records with SERVICE equal to SIPT. At the MCC or TLWS type and enter the command:
`OP:OFR:FORM=5829;`

Result:

```
OP OFR FORM=5829 PF is printed followed by a printout
    of the QGCON/5829 office records.
```

2 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

3 Type and enter the RC/V form number.

```
17.27
```

Result:

```
Enter Database Operation R=Review,U=Update:
```

4 Type and enter the update command.

```
U
```

Result:

The GENERAL GSM/NGSM CONNECTIVITY page is displayed and the cursor is located at the GSM field.

5 Using the RC/V 17.27 form as a guide, type and enter the parameters.

- GSM - GSM number
- SERVICE TYPE - SIPT

6 Change each SM-2000 in the NON-GSM list to a NULL.

7 Enter the update command.

```
U
```

Result:

```
Updating...form updated
```

8 Repeat steps 5-7 for each GSM identified in step 1.

9 Type and enter the previous screen command.

```
<
```

Result:

The CM MODULE VIEWS page is displayed.

---

10 If desired, type and enter the quit command to exit the RC/V system.

```
Q
```

Result:

The RC/V session is terminated.

---

11 At the MCC or TLWS type and enter the backup command.

```
BKUP:ODD;
```

Result:

```
BKUP ODD   COMPLETED
```

---

12 Stop. You have completed this procedure.

END OF STEPS

Delete All SIP GQPH Pipes (RC/V 17.24)

Purpose

The QPH Pipe Assignment (RC/V 17.24) defines the GQPH pipe connecting the QPH or GQPH to the QLPS network.

When to use

Use this procedure to deprovision all GQPH pipes for SIPT service.

NOTE: This is not needed for a DRM/VCDX environment.

Related information

Refer to 235-118-258, Recent Change Reference for information on the RC/V system.

Refer to 235-080-100, Translations Guide, for information on Office Records (output reports).

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Admonishments

The GQPH's will revert to STBY mode on MCC page 118X,Y,X.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking - NAR feature software.

Required Information

- GSM number
- PSU Shelf
- Channel Group
- QLPS Network

Required Conditions

- All connectivity for SIP-T service to NGSMs has been deleted on RC/V 17.27

Procedure

1 Identify the QPHPIPE/5828 office records with SERVICE equal to SIPT. At the MCC or TLWS type and enter the command:

```
OP:OFR:FORM=5828;
```

Result:

```
OP OFR FORM=5828 PF is printed followed by a printout
    of the QGCON/5828 office records.
```

Make a note of the GQPH pipes that have SIP-T service.

2 Deactivate each GQPH pipe identified as having SIP-T service in step 1.

```
RMV:GQPHPIPE=a-b-c-d-e,ucl;
```

Result:

```
The RMV GQPHPIPE report is printed and the GQPH pipe
   is deactivated.
```

3 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

4 Type and enter the RC/V form number.

```
17.24
```

Result:

```
Enter Database Operation I=Insert,R=Review,D=Delete:
```

5 Type and enter the delete command.

```
D
```

Result:

The QPH PIPE ASSIGNMENT page is displayed and the cursor is located at the GLOBAL SM field.

6 Using the RC/V 17.24 form as a guide, type and enter the parameters for the GQPH pipe.

- *GLOBAL SM
- *PSU SHELF

- *CHANNEL GROUP
- *QLPS NETWORK

---

7 Enter the delete command.

`D`

Result:

`Deleting...form deleted`

---

8 Repeat steps 6-7 for each GQPH pipe identified in step 1.

---

9 Type and enter the previous screen command.

`<`

Result:

CM MODULES VIEW page is displayed.

---

10 If desired, type and enter the quit command to exit the RC/V system.

`Q`

Result:

The RC/V session is terminated.

---

11 Go back and attempt to read the RC/V 17.27 view for each SIP GSM. The read should fail because it should have been automatically deleted with the last GQPHPIPE on that GSM. If it still exists, delete it.

---

12 At the MCC or TLWS type and enter the backup command.

`BKUP:ODD;`

Result:

`BKUP ODD   COMPLETED`

---

13 Stop. You have completed this procedure.

END OF STEPS

□

Update All PHE2 & PH33 Channel Group Assignments (RC/V 22.16)

Purpose

The PSU2 Protocol Handler Channel Group Assignments view (22.16) defines channel groups for protocol handlers equipped on a shelf. A channel group is a logical identifier for a protocol handler.

When to use

Use this procedure to unassign channel groups from PHE2 and PH33 hardware types.

Related information

Refer to 235-118-258, Recent Change Reference, for information on the RC/V system.

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking - NAR feature software.
- None

Required Information

- None

Required Conditions

- All SIP-T processor groups have been deleted on RC/V 33.16
- All IP processors that were used for SIP PHs have been deleted on RC/V 33.1
- All GQPHs have been deleted on RC/V 17.24
  NOTE: This is not needed for DRM/VCDX environment.
- The CRIT PSU (critical PSU2) must be set to N on RC/V 22.2.

Procedure

1 Type and enter the RC/V form number.

`22.16`

Result:

`Enter Database Operation R=Review,U=Update:`

2 Type and enter the update command.

`U`

Result:

Screen 1 of the PSU PROTOCOL HANDLER CHANNEL GROUP ASSIGNMENTS page is displayed and the cursor is located at the SM field.

3 Using the RC/V 22.16 form as a guide, type and enter the indicated values for each field on screen 1.

- *SM - Switching Module (1-192)
- *PSU - Packet Switch Unit (0)
- *PSU SHELF - PSU2 shelf number (0-4)
- *VIRTUAL - Virtual PSU2 shelf (Y/N)

4 Proceed to screen 2.

`2`

Result:

Screen 2 of the PSU PROTOCOL HANDLER CHANNEL GROUP ASSIGNMENTS page appears.

5 Using screen 2 of RC/V 22.16 as a guide type and enter NULL in the PH TYPE field for each channel group with a PH TYPE set to PHE2 or PH33 and GRP TYPE set to NULL.

6 Enter the update command.

`U`

Result:

`Updating...form updated`

7 Repeat steps 3-6 for each provisioned PSU shelf (0 thru 4) on each SIP GSM.

8 Type and enter the previous screen command.

`<`

Result:

ISDN -- EQUIPMENT VIEWS page is displayed.

9 If desired, type and enter the quit command to exit the RC/V system.

`Q`

Result:

The RC/V session is terminated.

10 At the MCC or TLWS type and enter the backup command.

`BKUP:ODD;`

Result:

```
BKUP ODD   COMPLETED
```

11 Stop. You have completed this procedure.

END OF STEPS

Delete All SIP Parameter Sets (RC/V 5.82)

Purpose

The SIP-T Parameters Definition view (RC/V 5.82) is used to delete a SIP parameter set for the packet trunking application.

When to use

Use this procedure to delete all SIP parameter set.

Related information

Refer to 235-118-258, Recent Change Reference for information on the RC/V system.

Refer to 235-080-100, Translations Guide, for information on Office Records (output reports).

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking - NAR feature software.

Required Information

- None.

Required Conditions

- All SIP-T packet groups have been deleted on RC/V 5.71

Procedure

1 Identify SIP parameter sets in the SIPTPARM/5882 office records. At the MCC or TLWS type and enter the command:

```
OP:OFR:FORM=5882;
```

Result:

```
OP OFR FORM=5882 PF is printed followed by a printout
    of the SIPTPARM/5882 office records.
```

2 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

3 Type and enter the RC/V form number.

`5.82`

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

4 Type and enter the delete command.

`D`

Result:

The SIP-T PARAMETERS DEFINITION page is displayed and the cursor is located at the PARM SET NAME field.

NOTE: You cannot delete the DEFAULT parameter because it will be removed when you delete in RC/V 5.80 later.

5 Type and enter the *PARM SET NAME field value identified in step 1.

6 Enter the delete command.

`D`

Result:

```
Deleting...form deleted
```

7 Repeat step 4-6 for every SIP parameter set identified in step 1.

8 Type and enter the previous screen command.

`<`

Result:

The TRUNK VIEWS page is displayed.

9 If desired, type and enter the quit command to exit the RC/V system.

`Q`

Result:

The RC/V session is terminated.

10 At the MCC or TLWS type and enter the backup command.

`BKUP:ODD;`

Result:

```
BKUP ODD   COMPLETED
```

11 Stop. You have completed this procedure.

END OF STEPS

Delete All SIP PSTN Interworking Parameter Sets (RC/V 5.83)

Purpose

The SIP PSTN Interworking Parameter Set view (RC/V 5.83) is used to delete a SIP PSTN interworking parameter set for the packet trunking application.

When to use

Use this procedure to delete all SIP PSTN interworking parameter sets .

Related information

Refer to 235-118-258, Recent Change Reference for information on the RC/V system.

Refer to 235-080-100, Translations Guide, for information on Office Records (output reports).

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR6 software release or later with the SIP for Packet Trunking - NAR feature software.

Required Information

- None.

Required Conditions

- All SIP-T packet groups have been deleted on RC/V 5.71

Procedure

1 Identify SIP PSTN interworking parameter sets in the SIPPSTN/5893 office records. At the MCC or TLWS type and enter the command: `OP:OFR:FORM=5893;`

Result:

```
OP OFR FORM=5893 PF is printed followed by a printout
    of the SIPPSTN/5893 office records.
```

2 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

3 Type and enter the RC/V form number.

```
5.83
```

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

4 Type and enter the delete command.

```
D
```

Result:

The SSIP PSTN INTERWORKING PARAMETER SET page is displayed and the cursor is located at the SET NAME field.

5 Type and enter the *SET NAME field value identified in step 1.

6 Enter the delete command.

```
D
```

Result:

```
Deleting...form deleted
```

7 Repeat step 4-6 for every SIP PSTN interworking parameter set identified in step 1.

8 Type and enter the previous screen command.

```
<
```

Result:

The TRUNK VIEWS page is displayed.

9 If desired, type and enter the quit command to exit the RC/V system.

`Q`

Result:

The RC/V session is terminated.

10 At the MCC or TLWS type and enter the backup command.

`BKUP:ODD;`

Result:

```
BKUP ODD   COMPLETED
```

11 Stop. You have completed this procedure.

END OF STEPS

Delete All SIP Global SMs (RC/V 5.80)

Purpose

The SIP Global SM view (RC/V 5.80) is used to insert, review, update, and delete the SIP Global Switching module.

When to use

Use this procedure to delete the SIP Global SM.

Related information

Refer to 235-118-258, Recent Change Reference for information on the RC/V system.

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking - NAR feature software.

Required Information

- GSM number

Required Conditions

- All SIP-T parameter sets except the DEFAULT set have been deleted on RC/V 5.82
- All SCTP far endpoints have been deleted on RC/V 33.21
- All SIP-T processor groups have been deleted on RC/V 33.16
- All SIP-T endpoint parameters have been deleted on RC/V 33.18
- All SCTP association parameter sets have been deleted on RC/V 33.20
- All GQPH pipes with SIP-T service have been deleted on RC/V 17.24

Procedure

1 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

2 Type and enter the RC/V form number.

```
5.80
```

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

3 Type and enter the delete command.

```
D
```

Result:

The SIP GLOBAL SM page is displayed and the cursor is located at the GSM field

4 Using the RC/V 5.80 form as a guide, type and enter the parameters.

- *GSM- GSM number

5 Enter the delete command.

```
D
```

Result:

```
Deleting...form deleted
```

6 Type and enter the previous screen command.

```
<
```

Result:

The TRUNK VIEWS page is displayed.

7 If desired, type and enter the quit command to exit the RC/V system.

```
Q
```

Result:

The RC/V session is terminated.

8 At the MCC or TLWS type and enter the backup command.

```
BKUP:ODD;
```

Result:

```
BKUP ODD   COMPLETED
```

9 Stop. You have completed this procedure.

END OF STEPS

Delete Protocol Handler (RC/V 22.4)

Purpose

The Equipment Subunit Protocol Handler view (22.4) defines the hardware for the protocol handlers equipped on the shelf.

When to use

Use this procedure to degrow a PH in the switch.

Related information

Refer to 235-118-258, Recent Change Reference for information on the RC/V system.

Refer to 235-105-331, Hardware Change Procedures - Degrowth for information on PH degrowth.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking - NAR feature software.
- The PHs must be installed in the shelf of the PSU2.

Required Information

- Switching Module (SM) Number
- Packet Switching Interface Unit Number
- Packet Switching Unit Shelf Number
- Position

Required Conditions

- The CRIT PSU (critical PSU2) must be set to N on RC/V 22.2.

Procedure

1 Refer to the growth procedures documentation.

Reference:

235-105-331, Hardware Change Procedures - Degrowth

END OF STEPS

Delete SIP Signaling from an Existing SIP Network

Introduction

This section contains additional procedures that might be required for deprovisioning all or part of the SIP signaling network.

List of procedures

Below are additional steps for deprovisioning the SIP application. This section assumes the reader is knowledgeable about the RC/V system and the reading of office records. More procedural information can be found in the links that are listed within the steps.

☐

Delete SIP PHE2 from Existing Duplex SIP IP Processor Group

Purpose

This procedure is used to delete a SIP PHE2 from an existing duplex SIP IP processor group.

When to use

Use this procedure to delete a SIP PHE2 from an existing duplex SIP IP processor group.

Related information

Refer to 235-118-258, Recent Change Reference for information on the RC/V system.

Refer to 235-105-331, Hardware Change Procedures - Degrowth for information on degrowth of the PHE2.

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking - NAR feature software.

Required Information

- SM, PSU, and PCR GRP (RC/V 33.16).
- Shelf and CHGRP of PHE2 to be removed from the SIP IP Processor group.

Required Conditions

- None

Procedure

1 Use the OP:STATUS,SERV command to determine whether the PHE2 to be deleted is currently the SERVING PH in the SIP-T processor group. If it is, use the SW:SERV:PCRGRP command to make it NON-SERVING.

NOTE: During a SW: (switch), transient calls will be dropped.

2 Use the RMV:PSUPH command to put the PHE2 manually OOS.

3 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

4 Update RC/V 33.16 to blank out the shelf/channel group/position/ethernet link (PHE LINK) data for the PH to be deleted.

This should automatically result in a deletion of RC/V 33.4 for the affected channel group (while the IP interface remains on 33.4 for the other channel group in the processor group), it should not be necessary/possible to do the 33.4 deletion manually.

5 Delete RC/V 33.1 for the channel group being removed from the processor group.

6 On RC/V 22.16 for the PSU shelf, ***verify*** that the GRP TYPE of the affected channel group is NULL.

On RC/V 22.16 screen 2 (or 3), for the appropriate row and group number, blank out the PH TYPE, AUTO ASSIGN, and RMK fields.

7 At the MCC or TLWS type and enter the backup command.

```
BKUP:ODD;
```

Result:

```
BKUP ODD   COMPLETED
```

8 Execute the Hardware Degrowth Procedures for the PHE2 and Ethernet Link.

Refer to 235-105-331, Hardware Change Procedures - Degrowth.

9 Stop. You have completed this procedure.

END OF STEPS

Delete an Existing SIP Call-Processing SM

Purpose This procedure is used to delete a SIP call-processing SM.

When to use Use this procedure to delete an existing SIP call-processing SM.

Related information Refer to 235-118-258, Recent Change Reference for information on the RC/V system.

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Admonishments None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking - NAR feature software.

Required Information

- Call Processing SM number and SIP GSM number

Required Conditions

- None

Procedure

1 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

2 On RC/V 5.81, update the ROUTE field for the desired SIP CP SMs to N, so that new calls will not be routed to those SMs.

3 Wait long enough for existing calls that are being handled by those SMs to terminate.

4 Update RC/V 5.81, blanking out the SIP CP SMs to be removed.

If there are still active SIP-T calls on the SM(s), the transaction will fail with an error message indicating that there are still active calls on the SIP CP SM. Wait, and try again, until the transaction succeeds.

5 At the MCC or TLWS type and enter the backup command.

```
BKUP:ODD;
```

Result:

```
BKUP ODD   COMPLETED
```

6 Stop. You have completed this procedure.

END OF STEPS

Delete Packet Trunking to Another Office

Purpose

This section contains the procedure for deleting packet trunking to another office.

When to use

Use this procedure to delete packet trunking to another office.

Related information

Refer to 235-118-258, Recent Change Reference for information on the RC/V system.

Refer to 235-080-100, Translations Guide, for information on Office Records (output reports).

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking - NAR feature software.

Required Information

- Packet Trunk Group number

Required Conditions

- None

Procedure

1 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

2 Delete existing routing/screening/digit analysis data that points to the trunk group of the packet group being deleted;

- Examine the MCRTIDX/5303 and RTIDX/5304 office records to find the route index(es) assigned to the trunk group associated with the packet group to be deleted.
- Update RC/V 10.2 and 10.4 to remove the trunk group from any route index(es) assigned to it, to stop new outgoing calls from being routed to it.

3 Coordinate with the far office, so that it also removes the packet trunk group from its route lists, to stop incoming calls from the far office.

4 Wait long enough to let existing stable calls terminate normally.

5 On RC/V 5.1, delete the trunk group associated with the packet group to be deleted.

The "Delete All SIP Packet Trunk Groups (RC/V 5.1)" (6-20) procedure can be used to delete ***all*** SIP Packet Trunk Groups, or in this case, just delete ***the specific*** SIP Packet Trunk Group.

6 Read RC/V 5.71 for the packet group and record the ASSOC SET NAME or UDP PATH that supports the packet group, then delete the packet group on RC/V 5.71.

The "Delete All SIP Packet Groups (RC/V 5.71)" (6-23) procedure can be used to delete ***all*** SIP Packet Groups, or in this case, to delete ***the specific*** SIP Packet Group.

7 If a UDP path was recorded in the previous step, and the UDP path to the far office will not be used again, delete the UDP path on RC/V 33.24.

8 If an ASSOC SET NAME was recorded in two step previous, and the association set to the far office will not be used again, execute the "Delete Association Set to Another Office" (6-97) procedure.

9 At the MCC or TLWS type and enter the backup command.

`BKUP:ODD;`

Result:

`BKUP ODD   COMPLETED`

10 Stop. You have completed this procedure.

END OF STEPS

Delete Association Set to Another Office

Purpose

This section defines the procedure for deleting an association set to another office.

When to use

Use this procedure to delete an association set to another office.

Related information

Refer to 235-118-258, Recent Change Reference for information on the RC/V system.

Refer to 235-080-100, Translations Guide, for information on Office Records (output reports).

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking - NAR feature software.

Required Information

- Association Set name

Required Conditions

- None

Procedure

1 Examine the PKTGRP/5219 office records to verify that the association set to be deleted is not assigned to any packet group. If it is, execute the “Delete Packet Trunking to Another Office” (6-94) procedure.

2 Use the OP:STATUS,SCTP,ASSOCSET command to determine the status of all the associations in the association set, and use the RMV:SCTP,ASSOC command to place all the associations in the CLOSED MAN state.

3 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

4 On RC/V 33.23, delete the association set.

The " Delete All SCTP Association Sets (RC/V 33.23)" (6-32) procedure can be used to delete ***all*** SCTP Association Sets. In this instance, to delete ***the specific*** SCTP Association Set.

5 At the MCC or TLWS type and enter the backup command.

```
BKUP:ODD;
```

Result:

```
BKUP ODD   COMPLETED
```

6 Stop. You have completed this procedure.

END OF STEPS

Delete SCTP Associations from Existing Association Set

Purpose

This section contains the procedure for deleting SCTP associations from an existing association set.

When to use

Use this procedure to delete an SCTP association from an existing association set.

Related information

Refer to 235-118-258, Recent Change Reference for information on the RC/V system.

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking - NAR feature software.

Required Information

- Association Set name
- Association numbers

Required Conditions

- None

Procedure

1 Use the RMV:SCTP,ASSOC command to put the association(s) that are to be removed from the association set in the CLOSED MAN state.

Note that any stable SIP-T calls that were established over the affected associations will no longer be able to signal to the far end, and therefore will not be able to terminate normally, so it would be best to do this during a low-traffic period. There is no way to reassign the calls to another association, nor is there any way to drain calls on a particular association within an association set.

---

2 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

---

3 Update RC/V 33.23, blanking out the association(s) to be deleted from the set.

Note: At least one association must remain in the set, otherwise execute the "Delete Association Set to Another Office" (6-97) procedure to delete the existing association set.

---

4 At the MCC or TLWS type and enter the backup command.

```
BKUP:ODD;
```

Result:

```
BKUP ODD   COMPLETED
```

---

5 Stop. You have completed this procedure.

END OF STEPS

Delete Existing SCTP Association to SCTP Far Endpoint

Purpose

This section contains the procedure for deleting existing SCTP associations to SCTP far endpoints.

When to use

Use this procedure to delete an existing SCTP association to an SCTP far endpoint.

Related information

Refer to 235-118-258, Recent Change Reference for information on the RC/V system.

Refer to 235-080-100, Translations Guide, for information on Office Records (output reports).

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking - NAR feature software.

Required Information

- Association number

Required Conditions

- None

Procedure

1 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

2 Type and enter the RC/V form number.

`33.22`

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

3 Type and enter the review command.

`R`

Result:

SCTP ASSOCIATION page is displayed and the cursor is located at ASSOC NUMBER field

4 Using the RC/V 33.22 form as a guide, type and enter the appropriate values for each field.

- *ASSOC NUMBER

5 Enter the review command.

`R`

Result:

View is updated.

6 Determine if the association is assigned to an association set. If it is, delete the SCTP association from the association set using the "Delete SCTP Associations from Existing Association Set" (6-99) procedure.

7 Execute the following command to verify the status of all provisioned associations:

`OP:STATUS,SCTP,ASSOC=XXXX;`

Where:

XXXX = association number

Result:

The "OP STATUS SCTP ASSOC =XXXX" output report is printed, with the status of the association, and it's Near and Far Endpoint names.

8 Verify that the association is in the CLOSED MAN state. This is the only state it should be in when it does not belong to an association set.

Type and enter the MML command:

```
RMV:SCTP:ASSOC=XXXX;
```

9 Record the near and far endpoints of the association from the report generated in step 7.

10 The" Delete All SCTP Associations (RC/V 33.22)" (6-35) procedure is usually used to delete ***all*** SCTP Associations, however, in this case, use the procedure to delete ***an individual*** SCTP Association.

11 Examine the SCTPASSC/5889 office records to determine whether the far endpoint of the deleted association is used by any other associations.

If the far endpoint is still used by other associations, ***Stop.*** You have completed this procedure.

12 The "Delete All SCTP Far Endpoints (RC/V 33.21)" (6-38) procedure is usually used to delete ***all*** SCTP Far Endpoints, however, in this case, use the procedure to delete ***one*** SCTP Far Endpoint.

13 Type and enter the RC/V form number.

```
33.19
```

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

14 Type and enter the review command.

```
R
```

Result:

SCTP NEAR ENDPOINT DEFINITION page is displayed and the cursor is located at NEAR ENDPOINT NAME field.

15 Using the RC/V 33.19 form as a guide, type and enter the appropriate values for each field.

- *NEAR ENDPOINT NAME

16 Enter the review command and record the processor group for the near endpoint.

```
R
```

Result:

```
The GSM and processor group for the NEAR ENDPOINT
   NAME listed.
```

17 Type and enter the previous screen command.

```
<
```

Result:

INTERNET PROTOCOL VIEWS page is displayed.

18 Type and enter the RC/V form number.

```
33.16
```

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

19 Type and enter the review command.

```
R
```

Result:

PROCESSOR GROUP page is displayed and the cursor is located at PCR GRP field

20 Using the RC/V 33.16 form as a guide and the data determined in Step 16, type and enter the appropriate values for each field.

- *PCR GRP
- *SM - SIP GSM number

21 Enter the review command.

`R`

Result:

`The view is refreshed with the channel group information.`

22 Determine the SM, PSU shelf, and channel group(s) assigned to the processor group.

23 Type and enter the previous screen command.

`<`

Result:

INTERNET PROTOCOL VIEWS page is displayed.

24 Type and enter the RC/V form number.

`33.4`

Result:

`Enter Database Operation I=Insert,R=Review,U=Update, D=Delete:`

25 Type and enter the review command.

`R`

Result:

Screen 1 of the ETHERNET INTERNET PROTOCOL (IP) INTERFACE ASSIGNMENT page is displayed and the cursor is located at the SM field.

26 Using screen 1 of the RC/V 33.4 form as a guide and the data recorded in Step 22, type and enter the parameters.

- *SM - Switching Module number
- *PSU - Packet Switching Unit number
- *SHELF - PSU shelf number
- *CHANNEL GROUP - Channel group position

27 Proceed to screen 2.

`2`

Result:

Screen 2 is displayed and the cursor is located at the GATEWAY IP ADDR1 field.

28 On screen 2 of form 33.4, get the Ethernet IP interface name.

29 Type and enter the previous screen command.

<

Result:

The INTERNET PROTOCOL VIEWS page is displayed.

30 Attempt to read RC/V 33.3 with the keys being the IP interface name and the IP address(es) of the far endpoint, recorded in step 28.

- If the read succeeds, for each IP address, delete the 33.3 view, since the explicit route to the far endpoint IP address is no longer needed.
- If the read fails, that IP address was routing more generally according to subnet, and there is no provisioned IP route to be deleted.

31 If desired, type and enter the quit command to exit the RC/V system.

`Q`

Result:

The RC/V session is terminated.

32 At the MCC or TLWS type and enter the backup command.

`BKUP:ODD;`

Result:

`BKUP ODD   COMPLETED`

33 Stop. You have completed this procedure.

END OF STEPS

☐

Delete SCTP Near Endpoint

Purpose This section contains the procedures for deleting an SCTP near endpoint.

When to use Use this procedure to delete an SCTP near endpoint

Related information Refer to 235-118-258, Recent Change Reference for information on the RC/V system.

Admonishments None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking - NAR feature software.

Required Information

- SCTP Near Endpoint name

Required Conditions

- No SCTP associations provisioned on endpoint, endpoint in GROW STATE

Procedure

1 Select and prepare terminal for RC/V activities.

Reference:

“Select and Prepare Terminal” (5-11)

2 Delete the SCTP near endpoint for the processor group using the procedure “Delete All SCTP Near Endpoints (RC/V 33.19)” (6-41). This procedure can be used to delete ***all*** SCTP Near Endpoints, or in this case, to delete ***an individual*** SCTP Near Endpoint.

3 At the MCC or TLWS type and enter the backup command.

```
BKUP:ODD;
```

Result:

```
BKUP ODD   COMPLETED
```

4 Stop. You have completed this procedure.

END OF STEPS

Delete Existing SIP PHE2 IP Processor Group

Purpose

This section contains the procedure for deleting an existing SIP PHE2 IP processor group.

When to use

Use this procedure to delete an existing SIP PHE2 IP processor group.

Related information

Refer to 235-118-258, Recent Change Reference for information on the RC/V system.

Refer to 235-080-100, Translations Guide, for information on Office Records (output reports).

Refer to the 235-600-700/750, Input/Output Messages manuals for additional information on the Input and Output commands.

Refer to 235-105-331, Hardware Change Procedures - Degrowth for information on degrowth of the PHE2.

Admonishments

None.

Before you begin

Required Tools

- Supplemental Trunk Line Workstation (STLWS) or Recent Change and Verify (RC/V) terminal
- Access to RC/V Menu Interface

Required Material

- The *5ESS*® switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking - NAR feature software.

Required Information

- GSM Number
- Processor Group number

Required Conditions

- Examine the SCTPNEPD/5886 office records to verify that there is no SCTP near endpoint assigned to the processor group. If there is one, first execute the "Delete SCTP Near Endpoint" (6-107) procedure.
- All UDP paths on the processor group to be deleted have been deleted (check UDPPATH/5891 office records, use RC/V 33.24 to delete selected UDP paths, similar to "Delete All SIIP UDP Paths" procedure, but only for UDP paths on the processor group to be deleted).

Procedure

1 Select and prepare terminal for RC/V activities.

Reference:

"Select and Prepare Terminal" (5-11)

2 Type and enter the RC/V form number.

```
33.16
```

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

3 Type and enter the review command.

```
R
```

Result:

PROCESSOR GROUP page is displayed and the cursor is located at PCR GRP field

4 Using the RC/V 33.16 form as a guide, type and enter the indicated values for each field.

- *PCR GRP
- *SM - SIP GSM number

5 Determine the SM, PSU shelf, and channel group(s) assigned to the processor group.

6 Type and enter the previous screen command.

<

Result:

INTERNET PROTOCOL VIEWS page is displayed.

7 Type and enter the RC/V form number.

`33.4`

Result:

```
Enter Database Operation I=Insert,R=Review,U=Update,
   D=Delete:
```

8 Type and enter the review command.

`R`

Result:

Screen 1 of the ETHERNET INTERNET PROTOCOL (IP) INTERFACE ASSIGNMENT page is displayed and the cursor is located at the SM field.

9 Using screen 1 of the RC/V 33.4 form as a guide, type and enter the parameters.

- *SM - Switching Module number
- *PSU - Packet Switching Unit number
- *SHELF - PSU shelf number
- *CHANNEL GROUP - Channel group position

10 On screen 1 of form 33.4, record the Ethernet IP INTERFACE NAME.

11 Type and enter the previous screen command.

<

Result:

The INTERNET PROTOCOL VIEWS page is displayed.

12 Dump the IPPING/5884 office records. At the MCC or TLWS type and enter the command:

```
OP:OFR:FORM=5884;
```

Result:

OP:OFR:FORM=5884 PF is printed followed by a printout of the IPPING/5884 office records.

13 Determine if there are IP Ping parameters assigned to the Ethernet IP interface. If there are, delete ***only the selected*** records using the "Delete All Router Pinging (RC/V 33.17)" (6-50) procedure.

14 Delete ***only the selected*** processor group using the "Delete All SIP Processor Groups (RC/V 33.16)" (6-63) procedure.

15 Delete IP processor assignment for all channel groups in the processor group. See 'Delete IP processor assignment to PH' procedure.

16 For the PSU shelf of each channel group in the deleted processor group, verify that the GRP TYPE is NULL, and update the PH TYPE from PHE2 to NULL using the procedure "Update All PHE2 & PH33 Channel Group Assignments (RC/V 22.16)" (6-75) .

17 At the MCC or TLWS type and enter the backup command.

```
BKUP:ODD;
```

Result:

```
BKUP ODD   COMPLETED
```

18 Execute the Hardware Degrowth procedures for each PHE2 and Ethernet Link in the processor group.

Refer to 235-105-331, Hardware Change Procedures - Degrowth.

19 Stop. You have completed this procedure.

END OF STEPS

7 Maintenance Considerations

Overview

Purpose

The purpose of this chapter is to provide the routine and corrective maintenance procedures that are unique to the Session Initiation Protocol (SIP) feature. Please refer to the *Routine Operations and Maintenance Procedures*, 235-105-210, *Corrective Maintenance Procedures*, 235-105-220, and *System Recovery*, 235-105-250, for additional support.

This chapter is subdivided into three subsections: Routine Maintenance, Corrective Maintenance, and Performance Monitoring.

When system faults occur in a Global Switching Module (GSM), maintenance software automatically attempts to recover from the failure or at least minimize the impact on system operation. If fault recovery cannot correct the problems, look for offnormal conditions among the following: associated Packet Switching Units (PSUs) and/or Protocol Handlers (PHs), Stream Control Transmission Protocol (SCTP) endpoints, SCTP heartbeat, SCTP associations and/or sets, provisioned processor groups (PCRGRPs), LLE2 paddleboard, Ethernet Links, General Quad Link Packet Switch (QLPS) Protocol Handler (GQPH) QPipes, general message transport, and intermediate network equipment such as an external router. In the case of problems with the intermediate network equipment, refer to troubleshooting documentation associated with that product. This chapter provides direction for all other problems listed.

Beyond fault recovery, in many cases, the use of SIP for Packet Trunking performance monitoring will provide the means to prevent SIP for Packet Trunking error conditions from occurring in the office. Corollary to this assumption of performance monitoring for both the near-end and far-end offices, most error conditions that occur will probably involve the intermediate network equipment such as the edge switch or the router. If there are no other observable office conditions such as SIP for Packet Trunking based audits, asserts, Machine-Detected Interoffice Irregularity (MDII) reports, error reports, or Master Control Center (MCC) status indicators, check the intermediate network equipment and the far office for problems.

□

Routine Maintenance

This section contains maintenance procedures to perform routine operations and maintenance on the *5ESS*® switch. The following subsections are included: Routine SCTP Status Requests, SCTP Testing, and Configuration Data Collection.

Note: The routine exercise (REX) currently does not apply to PHE2s. Follow *Routine Exercise Procedures,* 235-105-210, for using REX on PH33s.

□

Collect Data from Office Records

Purpose

This procedure provides the steps to obtain data that is required in procedures throughout this chapter.

The following office records and associated data can be obtained from this procedure:

- Packet Group (Office Record: 5219)
- GSM (Office Record: 5880)
- Provisioned Processor Groups (Office Record: 5883)
- SCTP Near Endpoint (Office Record: 5886)
- SCTP Association Name (Office Record: 5889)
- SCTP Association Set Name (Office Record: 5890)
- IP Processor Assignment (Office Record: 5987)
- IP Interface Assignment (Office Record: 5988)
- Ethernet IP Interface Assignment (Office Record: 5995)
- UDP Path (Office Record: 5891)
- SIP PSTN Interworking Parameter Set (Office Record: 5893)
- Inter-Nodal Trunk Group Identifier (Office Record: 5894)

When to use

Use this procedure for identifying GSM and SCTP-based endpoints, associations, or association sets, and UDP Paths. This procedure is listed as a prerequisite in each procedure that requires the identified data.

Related information

Refer to the *Input Messages Manual*, 235-600-700, and the *Output Messages Manual*, 235-600-750, for additional information.

Admonishments

None.

Before you begin

Required Tools

- Use of the Master Control Center (MCC) or Supplementary Trunk Line Workstation (STLWS).

Required Material

- The switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking feature software.

Required Information

- None.

Conditions

- None.

Procedure

1 Identify Packet Group Numbers by outputting office record 5219.

```
OP:OFR:FORM=5219;
```

Result:

Output of Office Record 5219 prints.

2 Identify GSMs by outputting office record 5880.

```
OP:OFR:FORM=5880;
```

Result:

Output of Office Record 5880 prints.

3 Identify provisioned processor groups by outputting office record 5883.

```
OP:OFR:FORM=5883;
```

Result:

Output of Office Record 5883 prints.

4 Identify SCTP Near Endpoints by outputting office record 5886.

```
OP:OFR:FORM=5886;
```

Result:

Output of Office Record 5886 prints.

5 Identify SCTP Association Names by outputting office record 5889.

```
OP:OFR:FORM=5889;
```

Result:

Output of Office Record 5889 prints.

6 Identify SCTP Association Set Names by outputting office record 5890.

```
OP:OFR:FORM=5890;
```

Result:

Output of Office Record 5890 prints.

7 Identify IP Routing to Interface by outputting office record 5987.

```
OP:OFR:FORM=5987;
```

Result:

Output of Office Record 5987 prints.

8 Identify IP Processor Interface Assignment by outputting office record 5988.

```
OP:OFR:FORM=5988;
```

Result:

Output of Office Record 5988 prints.

9 Identify IP Provisioning by outputting office record 5989.

```
OP:OFR:FORM=5989;
```

Result:

Output of Office Record 5989 prints.

10 Identify Ethernet IP Interface Assignment by outputting office record 5995.

```
OP:OFR:FORM=5995;
```

Result:

Output of Office Record 5995 prints.

11 Identify UDP Paths by outputting office record 5891.

```
OP:OFR:FORM=5891;
```

Result:

Output of Office Record 5891 prints.

12 Identify SIP PSTN interworking parameter sets by outputting office record 5893.

```
OP:OFR:FORM=5893;
```

Result:

Output of Office Record 5893 prints.

13 Identify packet group to service group (trunk group) associations by outputting office record 5894.

```
OP:OFR:FORM=5894;
```

Result:

Output of Office Record 5894 prints.

END OF STEPS

Request SCTP Near Endpoint Status

Purpose

This procedure requests:

- the status of a specific SCTP near endpoint,
- the status of a specific SCTP near endpoint and for all associations assigned to that near endpoint, or
- the status of all endpoints.

When to use

According to local practices for routine maintenance, use this procedure to obtain the status of an SCTP near endpoint.

Three procedures are provided to obtain:

1. General status of a specific near endpoint
2. Detailed status on a specific near endpoint
3. Status of all near endpoints

Related information

Refer to the *Input Messages Manual*, 235-600-700, and the *Output Messages Manual*, 235-600-750, for additional information.

Admonishments

None

Before you begin

Required Tools

- Use of the Master Control Center (MCC) or Supplementary Trunk Line Workstation (STLWS).

Required Material

- The switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking feature software.

Required Information

- Office Record 5886 to identify endpoint names of a minimum of one up to a maximum of all provisioned SCTP Near endpoints.

Required Conditions

- None required.

Procedure to Obtain General Status of Near Endpoint

1 Request a status report for a specific near endpoint:

`OP:STATUS,SCTP,NEAREPT=a;`

Where:

a = near endpoint name

Result:

Generates a status report of a specific SCTP near endpoint.

2 Check the result and take appropriate action:

- If the result is `NG - INVALID NEAR ENDPOINT NAME`, then output the associated office record:
  `OP:OFR:FORM=5886;`
  Refer to *Translation Guide (TG-5)*, 235-080-100, for obtaining office record numbers.
- If the result includes `NOSERV` or `OOS`, refer to the SIP for Packet Trunking Corrective Maintenance procedure, *Resolve SCTP Near Endpoint Problems*, 235-200-118, to resolve the problem.
- For all others, refer to the *Output Messages Manual*, 235-600-750, for additional information.

3 Stop. You have completed this procedure.

END OF STEPS

Procedure to Obtain Detailed Status of Near Endpoint

1 Request a detailed status report for a specific near endpoint:
`OP:STATUS,SCTP,NEAREPT=a,DETAIL;`

Where:

a = near endpoint name

Result:

Generates a status report of a specific near endpoint and for all associations assigned to that near endpoint.

2 Check the result and take appropriate action:

- If the result is `NG - INVALID NEAR ENDPOINT NAME`, then output the associated office record:
  `OP:OFR:FORM=5886;`
  Refer to *Translation Guide (TG-5)*, 235-080-100, for obtaining office record numbers.
- If the result is a status of `NOSERV` or `OOS`, refer to the SIP for Packet Trunking Corrective Maintenance procedure, *Resolve SCTP Near Endpoint Problems*, 235-200-118, to resolve the problem.
- For all others, refer to the *Output Messages Manual*, 235-600-750, for additional information.

3 Stop. You have completed this procedure.

END OF STEPS

Procedure to Obtain Status of All Near Endpoints

1 Request a status report for all near endpoints:
`OP:STATUS,SCTP,ALL;`

Result:

Generates a status report of all near endpoints.

2 Check the result and take appropriate action:

- If the result of any of the endpoints includes `NOSERV` or `OOS`, refer to the SIP for Packet Trunking Corrective Maintenance procedure,*Resolve SCTP Near Endpoint Problems*, 235-200-118, to resolve the problem.
- For all others, refer to the *Output Messages Manual*, 235-600-750, for additional information.

3 Stop. You have completed this procedure.

END OF STEPS

□

Request SCTP Association Status

Purpose

This procedure requests

- the status of a specific SCTP association,
- the status of a specific SCTP association including the path status, or
- the status of all associations.

When to use

According to local practices for routine maintenance, use this procedure to obtain the status of an SCTP association.

Related information

Refer to the *Input Messages Manual*, 235-600-700, and the *Output Messages Manual*, 235-600-750, for additional information.

Admonishments

None

Before you begin

Required Tools

- Use of the Master Control Center (MCC) or Supplementary Trunk Line Workstation (STLWS).

Required Material

- The switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking feature software.

Required Information

- Office Record 5889 for a listing of identified associations.

Required Conditions

- None required.

Procedure to Obtain Status of a Specific Association

1 Request a status report for a specific association:

```
OP:STATUS,SCTP,ASSOC=a;
```

Where:

a = association number

Result:

Generates a status report of a specific SCTP association.

2 Check the result and take appropriate action:

- If the result is `NG - INVALID ASSOCIATION ID`, then examine the associated office record:
  `OP:OFR:FORM=5889;`
  Refer to *Translation Guide (TG-5)*, 235-080-100, for obtaining office record numbers.
- If the result is a status of `CLOSED`, refer to *Resolve SCTP Association Problems*, 235-200-118, to resolve the problem.
- For all others, refer to the *Output Messages Manual*, 235-600-750, for additional information.

3 Stop. You have completed this procedure.

END OF STEPS

Procedure to Obtain Status Including Paths for a Specific Association

1 Request a status report for a specific association including the path status:

`OP:STATUS,SCTP,ASSOC=a,PATHS;`

Where:

a = association number

Result:

Generates a status report of a specific association including the path status.

2 Check the result and take appropriate action:

- If the result is `NG - INVALID ASSOCIATION ID`, then examine the associated office record:
  `OP:OFR:FORM=5889;`
  Refer to *Translation Guide (TG-5),* 235-080-100, for obtaining office record numbers.
- If the result is a status of `CLOSED`, then refer to the SIP for Packet Trunking Corrective Maintenance procedure, *Resolve SCTP Association Problems*, 235-200-118, to resolve the problem.
- For all others, refer to the *Output Messages Manual,* 235-600-750, for additional information.

3 Stop. You have completed this procedure.

END OF STEPS

Procedure to Obtain Status of All Associations

1 Request a status report for all associations:
`OP:STATUS,SCTP,ASSOC,ALL;`

Result:

Generates a status report of all associations.

2 Check the result and take appropriate action:

- If the result of any of the associations is a status of `CLOSED`, then refer to the SIP for Packet Trunking Corrective Maintenance procedure, *Resolve SCTP Association Problems*, 235-200-118, to resolve the problem.
- For all others, refer to the *Output Messages Manual,* 235-600-750, for additional information.

3 Stop. You have completed this procedure.

END OF STEPS

□

Request SCTP Association Set Status

Purpose

This procedure requests the status of each association in an SCTP association set.

When to use

According to local routine maintenance practices, use this procedure to obtain the status of each association in an SCTP association set.

Related information

Refer to the *Input Messages Manual*, 235-600-700 and the *Output Messages Manual*, 235-600-750, for additional information.

Admonishments

None

Before you begin

Required Tools

- Use of the Master Control Center (MCC) or Supplementary Trunk Line Workstation (STLWS).

Required Material

- The switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking feature software.

Required Information

- Office Record 5890 with a listing of identified association sets.

Required Conditions

- None required.

Procedure

1 Request a status report for each association in an association set:

```
OP:STATUS,SCTP,ASSOCSET=c;
```

Where:

c = Association set name

Result:

Generates a status report for each association in a specific association set.

2 Check the result and take appropriate action:

- If the result is `NG - INVALID ASSOCIATION SET`, then examine the associated office record:
  `OP:OFR:FORM=5890;`
  Refer to *Translation Guide (TG-5)*, 235-080-100, for obtaining office record numbers.
- For all others, refer to the *Output Messages Manual*, 235-600-750, for additional information.

3 Stop. You have completed this procedure.

END OF STEPS

Report Service Selection, Ping and Ethernet Link Status

Purpose

This procedure requests service selection, ping and Ethernet link status based on processor group or processor in processor group.

When to use

According to local routine maintenance practices, use this procedure to obtain the service selection, ping and Ethernet link status of a processor group.

There two ways to output the same report:

1. Request status based on processor group.
2. Request status based on PSU, shelf, and PH position number.

Related information

Refer to the *Input Messages Manual*, 235-600-700, and the *Output Messages Manual*, 235-600-750, for additional information.

Admonishments

None

Before you begin

Required Tools

- Use of Master Control Center (MCC) or Supplementary Trunk Line Workstation (STLWS).

Required Material

- The switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking feature software.

Required Information

- Office Record 5880 for provisioned GSMs.
- Office Record 5883 for provisioned processor groups.
- GSM number.
- PSU number, PSU shelf number, PH position number associated with identified processor group.

Required Conditions

- At least one serving SIP PH is in service when requesting based on processor in processor group.

Procedure Based on Processor Group Number

1 Request the service selection status of a processor group specifying a processor group:

`OP:STATUS,SERV,PCRGRP=a-b[&&b];`

Where:

a = Global switch module number

b = Processor group

Result:

Generates the service selection, ping and Ethernet link status report for a specific processor group or range of processor groups.

2 Stop. You have completed this procedure.

END OF STEPS

Procedure Based on PSU, Shelf, and PH Position

1 Request the service selection status of a processor in processor group:

`OP:STATUS,SERV,PSUPH=a-c-d-e;`

Where:

a = Global switch module number

c = Packet switch unit 2

d = PSU2 shelf number

e = PH position number

Result:

Generates the service selection, ping and Ethernet link status report for a specific processor.

2 Stop. You have completed this procedure.

END OF STEPS

Initiate SCTP Heartbeat

Purpose

This procedure requests that a manual SCTP heartbeat test be executed to evaluate the SCTP association path for

- each association in an association set,
- each association in an association set that serves the packet group,
- a specific association number or
- a specific destination address

When to use

According to local practices for routine maintenance, use this procedure for verifying connectivity of an SCTP association path or the transport layer to the far end.

Related information

Refer to the *Input Messages Manual*, 235-600-700, and the *Output Messages Manual*, 235-600-750, for additional information.

Admonishments

None.

Before you begin

Required Tools

- Use of Master Control center (MCC) or Supplementary Trunk Line Workstation (STLWS).

Required Material

- The switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking feature software.

Required Information

- Office Record 5889 for Association Data.
- Office Record 5890 for Association Set Data.
- Office Record 5219 for Packet Group Data.
- Destination IP address.

Required Conditions

- None required.

Procedure

1 Initiate a manual heartbeat test on a specific association set name:

`TST:PATH,SCTPHB,ASSOCSET=a;`

Where:

a = association set name

Result:

Generates an SCTP heartbeat report for each association in an association set in a specific association set. If the result is other than `COMPLETED: TEST PASSED`, then CONTINUE with this procedure. Otherwise, EXIT this procedure.

2 Check results and take appropriate action:

- If the result is `NG - INVALID ASSOCIATION SET NAME`, then examine the associated office record 5890.
  Refer to *Translation Guide (TG-5),* 235-080-100, for obtaining office record numbers.
- For all others, refer to the *Output Messages Manual,* 235-600-750, for additional information.

3 Stop. You have completed this procedure.

END OF STEPS

Procedure

1 Initiate a manual heartbeat test on a specific packet group:

`TST:PATH,SCTPHB,PKTG=b;`

Where:

b = Packet group

Result:

Generates an SCTP heartbeat report for each association in an association set that serves a specific packet group. If the result is other than `COMPLETED: TEST PASSED`, then CONTINUE with this procedure. Otherwise, EXIT this procedure.

2 Check results and take appropriate action:

- If the result is `NG - INVALID PACKET GROUP`, then examine the associated office record 5219.
  Refer to *Translation Guide (TG-5)*, 235-080-100, for obtaining office record numbers.
- For all others, refer to the *Output Messages Manual*, 235-600-750, for additional information.

3 Stop. You have completed this procedure.

END OF STEPS

Procedure

1 Initiate a manual heartbeat test on a specific association number:

`TST:PATH,SCTPHB,ASSOC=c;`

Where:

c = association number

Result:

Generates an SCTP heartbeat report for a specific association number. If the result is other than `COMPLETED: TEST PASSED`, then CONTINUE with this procedure. Otherwise, EXIT this procedure.

2 Check results and take appropriate action:

- If the result is `NG - INVALID ASSOCIATION ID`, then examine the associated office record 5889.
  Refer to *Translation Guide (TG-5)*, 235-080-100, for obtaining office record numbers.
- For all others, refer to the *Output Messages Manual*, 235-600-750, for additional information.

3 Stop. You have completed this procedure.

END OF STEPS

Procedure

1 Initiate a manual heartbeat test on a specific IP destination path or address:

`TST:PATH:SCTPHB,a,DIP=b;`

Where:

a = either ASSOCSET=d, PKTG=e or ASSOC=f;

d = association set

e = packet trunk group

f= Association

b = IP destination path or address

Result:

Generates an SCTP heartbeat report for a specific IP destination path or address. If the result is other than `COMPLETED: TEST PASSED`, then CONTINUE with the procedure. Otherwise, EXIT the procedure.

2 Check results and take appropriate action:

- Refer to the *Output Messages Manual*, 235-600-750, for additional information.

3 Stop. You have completed this procedure.

END OF STEPS

Initiate 105 Test Call

Purpose

This procedure requests an outgoing 105 test to a far-end 105 testline over the packet network for a SIP call. ***Note:*** This is only supported for SCTP.

When to use

According to local practices for routine maintenance, use this procedure to verify the quality of an entire end-to-end connection for a SIP call.

Related information

Refer to the *Input Messages Manual*, 235-600-700 and the *Output Messages Manual*, 235-600-750, for additional information. Since there are no Call Instance Codes (CICs) involved in SIP signaling, only option, Packet Trunk Group (PKTG) is allowed and not Packet Trunk Group Member Number (PKTGMN).

Admonishments

None

Before you begin

Required Tools

- Use of Master Control center (MCC) or Supplementary Trunk Line Workstation (STLWS).

Required Material

- The switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking feature software.

Required Information

- OPDN
- Office Record 5219 for Packet Group

Required Conditions

- None required.

Procedure

1 Request an outgoing test to a far-end 105 test line:

```
TST:PATH,OG105,PKTG=f,OPDN=d;
```

Where:

d = 105 test line DN for desired far-end office

f = Packet group

Result:

Check results and take appropriate action:

- If the result is `NG - INVALID PACKET GROUP`, then examine the associated office record 5219 for Packet Group. Refer to *Translation Guide (TG-5)*, 235-080-100, for obtaining office record numbers.
- For all others, refer to the *Output Messages Manual*, 235-600-750, for additional information.

2 Stop. You have completed this procedure.

END OF STEPS

Initiate Delay Test Call

Purpose

This procedure requests a delay test to a far-end loopback point over the packet network for a SIP call. ***Note:*** This is only supported for SCTP.

When to use

According to local practices for routine maintenance, use this procedure to verify loopback, delay, power and noise levels of a connection for a SIP call.

Related information

Refer to the *Input Messages Manual*, 235-600-700 and the *Output Messages Manual*, 235-600-750, for additional information. Since there are no CICs involved in SIP signaling, only option, PKTG is allowed and not PKTGMN.

Admonishments

None

Before you begin

Required Tools

- Use of Master Control center (MCC) or Supplementary Trunk Line Workstation (STLWS).

Required Material

- The switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking feature software.

Required Information

- Office Record 5219 for PKTG
- OPDN

Required Conditions

- None required.

Procedure

1 Request a delay test on a loopback testline:

`TST:PATH,DELAY,PKTG=f,OPDN=d;`

Where:

d = Loopback testline

f = Packet group

Result:

Check results and take appropriate action:

- If the result is “NG - INVALID PACKET GROUP”, then examine the associated office record 5219. Refer to *Translation Guide (TG-5)*, 235-080-100, for obtaining office record numbers.
- If the results are unacceptable, verify the provisioning and local connections.
- For all others, refer to the *Output Messages Manual*, 235-600-750, for additional information.

2 Stop. You have completed this procedure.

END OF STEPS

Request Utility Call Trace

Purpose

This procedure requests a utility call trace of the resources used by a SIP signaling call that is in progress to verify the signaling and bearer paths.

When to use

According to local practices for routine maintenance, use this procedure to conduct a call trace of a SIP signaling call.

Related information

Refer to the *Input Messages Manual*, 235-600-700 and the *Output Messages Manual*, 235-600-750, for additional information.

Admonishments

None

Before you begin

Required Tools

- Use of Master Control Center (MCC) or Supplementary trunk line workstation (STLWS).

Required Material

- The switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking feature software.

Required Information

- LTAG value can be derived from the output of a trace initiated at the circuit side of the connection or from the SIP header message output of the protocol trace.

Required Conditions

- SIP signaling call is in progress.

Procedure

1 Request a utility call trace:

```
TRC:UTIL,LTAG=a-b-c-d-e;
```

Where:

a = Global SM index

b = Processor group number

c = Global switch module number

d = Map Tag index

e = Terminal process number

Result:

The SIP signaling call trace is displayed on the MCC.

2 Stop. You have completed this procedure.

END OF STEPS

Procedure

1 Request a utility call trace by poking:

`132,2,465,LTAG X=a,b,c,d,e;`

Where:

a = Global SM index

b = Processor group number

c = Global switch module number

d = Map Tag index

e = Terminal process number

Result:

The SIP signaling call trace is displayed on the MCC.

2 Stop. You have completed this procedure.

END OF STEPS

Request Internet Protocol (IP) Configuration Data

Purpose

This procedure requests IP address configuration data to be displayed.

When to use

According to local routine maintenance practices, use this procedure to verify IP address configuration data. For example, if growing a SIP PH, check for IP address duplication against existing IP address configuration data.

Related information

Use Office Records 5987, 5988, and 5995 for off-line analysis. Refer to the *Input Messages Manual*, 235-600-700 and the *Output Messages Manual*, 235-600-750, for additional information.

Admonishments

An enormous amount of data may be printed on the Receive-only Printer (ROP). The use of any format could cause a degradation in switch service. You may want to list the office records that can be examined off-line. However, the printing of this data may be stopped using STP:IPCFG should system operation be impacted.

Before you begin

Required Tools

- Use of Master Control Center (MCC) or Supplementary Trunk Line Workstation (STLWS).

Required Material

- The switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking feature software.

Required Information

- Associated SM number
- IP Address
- Network ID

Required Conditions

- None required.

Procedure

1 Request IP address configuration data for an office:
`OP:IPCFG,OFFICE;`

Result:

Displays IP address configuration data for the office.

2 Stop. You have completed this procedure.

END OF STEPS

Procedure

1 Request IP address configuration data for a specific SM:

`OP:IPCFG,SM=a`

Where:

a = SM number

Result:

Displays IP address configuration data for a specific SM.

2 Stop. You have completed this procedure.

END OF STEPS

Procedure

1 Request configuration data for a specific IP address:

`OP:IPCFG,IPADR=b`

Where:

b = IP address

Result:

Displays configuration data for a specific IP address.

2 Stop. You have completed this procedure.

END OF STEPS

Procedure

1 Request IP address configuration data for a specific Network ID:

`OP:IPCFG,NID=c`

Where:

c = Network ID

Result:

Displays IP address configuration data for a specific Network ID.

2 Stop. You have completed this procedure.

END OF STEPS

Corrective Maintenance

Scope

This section contains SIP for Packet Trunking corrective maintenance procedures on the hardware and software of the *5ESS*® switch. These topics are covered: Problem Resolution, Problem Analysis, and Data

□

Resolve Protocol Handler Problems

Purpose This procedure identifies methods to restore `OOS` PHE2s and PH33s.

Related information Refer to the *Input Messages Manual*, 235-600-700 and the *Output Messages Manual*, 235-600-750, for additional information.

Admonishments None

Before you begin

Required Tools

- Use of Master Control Center (MCC)

Required Material

- The switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking feature software.

Required Information

- GSM number, PSU number, PSU shelf number

Required Conditions

- None required.

Procedure

1 Find the `OOS` channel group on the MCC which shows as backlit on the PSU shelf MCC page. Access the PSU 0 SHELF page:
`118X,Y,ZZZ`

Where:

X = Packet switch unit 2 shelf number

Y = PSU2 number

ZZZ = Global switch module number

When all provisioned PSU2 shelves have been checked and there are no more`OOS` channel groups, then STOP, you have finished this procedure.

2 If there are one or more unassigned channel groups, identify the PH type, choose an OOS PH of the same type to repair, and CONTINUE this procedure.

When interrogating unassigned channel groups, it is important to note the PH hardware type supporting the identified channel group and the appropriate OOS PH must be analyzed by the remainder of this procedure.

3 If the status of all equipped PHs is `OOSF`, an `ACT PSUCOM` does not exist. EXIT this procedure immediately, and refer to the System Recovery [5E14 and Later Software Releases] procedure, *PSU Duplex Failure,* 235-105-250, to resolve the duplex failed PSUCOM situation.

4 For the OOS PH, determine detailed status:

```
OP:CFGSTAT,SM=a,OOS,NOFE;
```

Where:

a = Global switch module number

Result:

- If the state is **OOS-MAN-RMV**, PROCEED to Step 5.
- If the state is **OOS-AUTO-RMV**, SKIP to Step 7.
- If the state is **OOS-MAN/AUTO-DGN**, SKIP to Step 8.
- If the state is **OOS-MAN-EX**, SKIP to Step 9.
- If the state is **OOS-AUTO/MAN-FLT**, SKIP to Step 10.
- If the state is **OOS-AUTO-TBLA**, SKIP to Step 11.
- If the state is **DGR**, SKIP to Step 12.

5 Restore the PHE2 or PH33:

```
RST:PSUPH=a-b-c-d;
```

Where:

a = Global switch module number

b = Packet switch unit 2 number

c = PSU2 shelf number

d = PHE2 number

6 Did the PHE2 or PH33 restore successfully to ACT?

- If **yes**, RETURN to Step 1
- If **no**, RETURN to Step 2

7 The PHE2 or PH33 has been automatically removed for the purpose of running diagnostics. Allow the diagnostic to be scheduled because a problem has been detected by fault recovery. After the diagnostics complete, RETURN to Step 1.

8 The PHE2 or PH33 is currently being diagnosed. When the diagnostics complete, RETURN to Step 1.

9 The PH33 is currently being routinely exercised. Because PH channel groups are unassigned, the routine exercise should be stopped immediately, then RETURN to Step 1 for re-evaluation.

```
STP:PSUPH=a-0-b-c;
```

Where:

a = Global switch module number

b = PSUCOM shelf number

c = Physical PH number

10 The PHE2 or PH33 is faulty. Find the most recent printed trouble location procedure (TLP) list on the ROP. For the PHE2 or PH33, refer to the Corrective Maintenance procedure, *Clear Diagnostic Failure in Hardware Units/Circuits*, 235-105-220.

After the repair is completed, including the attempted restoration of the PHE2 or PH33, is the PH successfully restored to ACT?

- If **yes**, RETURN to Step 1.
- If **no**, SEEK TECHNICAL ASSISTANCE according to local procedures for the affected PH.

If the channel groups on the target shelf are still unassigned and there are other OOS PHE2 or PH33 candidates that may still be successfully recovered, choose a new PHE2 or PH33 and RETURN to Step 2.

11 The PHE2 or PH33 is in a trouble analysis state. SEEK TECHNICAL ASSISTANCE according to local procedures for the affected PH.

12 If the PHE2 is degraded and a service selection change report is generated indicating that the Ethernet link is down, then determine the cause, for example, loose or faulty cable, and repair it, if possible. A corresponding `REPT PSELNK` report should have been generated.

However, it is possible that the LLE2 paddleboard is faulty. Refer to the SIP for Packet Trunking Corrective Maintenance procedure, *Resolve LLE2 Paddleboard Problems*, 235-200-118.

13 Stop. You have completed this procedure.

END OF STEPS

Resolve LLE2 Paddleboard Problems

Purpose

This procedure identifies methods for resolving LLE2 paddleboard problems.

When to use

In the course of resolving PH Problems, use this procedure when suspecting a problem with the LLE2 paddleboard associated with a PHE2 in the degraded state and associated Ethernet link is `OOS`.

Related information

Refer to the *Input Messages Manual*, 235-600-700 and the *Output Messages Manual*, 235-600-750, for additional information.

Admonishments

Before starting this procedure, read the general requirements for handling and care of circuit packs outlined in the System Maintenance procedure, *Circuit Pack Handling Procedures,* 235-105-110.

Before you begin

Required Tools

- Use of Master Control Center (MCC) or Supplementary Trunk Line Workstation (STLWS).

Required Material

- The switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking feature software.

Required Information

- PHE2 number in degraded state that is associated with an Ethernet link that is `OOS`.
- GSM number associated with the identified PHE2.
- PSU2 number associated with the identified PHE2.
- PSU2 shelf number containing the identified PHE2.

Required Conditions

- None required.

Procedure

1 At the identified PSU2 shelf, observe the OOS LED is lit on the LLE2 paddleboard which resides on the top half of the backplane of the PHE2 circuit pack (TN13). The LLE2 paddleboard plugs onto the upper pin field directly behind the TN13 circuit pack. If the OOS LED is not lit, then EXIT this procedure.

2 If the identified PHE2 is not `OOS`, remove it from service.

`RMV:PSUPH=a-b-c-d;`

Where:

a = Global switch module number

b = Packet switch unit 2 number

c = PSU2 shelf number

d = PHE2 number

3 At the PSU2 shelf, remove the corresponding LLE2 paddleboard.

4 Insert another LLE2 paddleboard in place of the removed paddleboard.

5 Restore the PHE2 associated with the replace LLE2 paddleboard.

`RST:PSUPH=a-b-c-d;`

Where:

a = Global switch module number

b = Packet switch unit 2 number

c = PSU2 shelf number

d = PHE2 number

Result:

Verify diagnostic phase 4 which tests the LLE2 paddleboard is All Tests Passed (ATP). Check results and take appropriate action:

- If diagnostics pass and the PHE2 restoral succeeds, then EXIT this procedure.
- If diagnostics fail at phase 4, repeat this procedure with another LLE2 paddleboard.
- If diagnostics pass but the PHE2 restoral fails, refer to the *Output Messages Manual*, 235-600-750, for additional information.

6 Stop. You have completed this procedure.

END OF STEPS

Resolve GQPH QPipe Problems

Purpose

This procedure identifies methods to restore `OOS` General QLPS (GQPH) QPipes.

Note:The Resolve GQPH QPipe Problems does not apply to DRM/VCDX.

Related information Refer to the *Input Messages Manual*, 235-600-700, and the *Output Messages Manual*, 235-600-750, for additional information.

Admonishments None

Before you begin

Required Tools

- Use of Master Control Center (MCC) or Supplementary trunk Line Workstation (STLWS).

Required Material

- The switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking feature software.

Required Information

- Office Record 5880 that lists the GSM associated with the identified QPipe.

Required Conditions

- None required.

Procedure

1 Request the status of the GQPH QPipe:

`OP:STATUS,GQPHPIPE,QPIPE=a-b-c-d-e;`

Where:

a = Global switch module number

b = Packet switch unit 2 number

c = PSU2 shelf number

d = GQPH channel group number

e = QLPS network number

Result:

Examine the output. If any QPipes are not in the `ACT` state, then CONTINUE with this procedure. Otherwise, EXIT this procedure.

- If the state is **OOS-DACT**, CONTINUE this procedure.
- If the state is **OOSF-CM**, SKIP to Step 3.
- If the state is **OOSF-PH**, SKIP to Step 4.

- If the state is **OOS-GQPHLB**, SKIP to Step 5.
- If the state is **OOS-LVL1-FRAME**, SKIP to Step 6.
- If the state is **OOS-LVL1-PATH**, SKIP to Step 10.
- If the state is **INIT**, SKIP to Step 12.
- If the state is **UNKNOWN**, SKIP to Step 13.

---

2 An `OOS-DACT` GQPH QPipe has been manually removed. Unless there is a valid reason for maintaining this status, restore the GQPH QPipe:

`RST:GQPHQPIPE=a-b-c-d-e;`

Where:

a = Global switch module number

b = Packet switch unit 2 number

c = PSU2 shelf number

d = GQPH channel group number

e = QLPS network number

Result:

Check results and take appropriate action:

- If the restore fails with the reason, `BOTH GQPH QPIPES MUST BE EQUIPPED`, then incomplete provisioning of the GQPH has occurred. Refer to the SIP for Packet Trunking Provisioning procedure, *Assign GQPH Pipe*, 235-200-118, to complete the growth of the second GQPH QPipe on the target GQPH channel group.
- If any other reason, RETURN to Step 1 and re-analyze the GQPH QPipe status.

---

3 An `OOSF-CM GQPH QPipe` has been removed because the parent QTMSLNK or TMSLNK hardware is `OOS`. Determine the parent hardware to be recovered by accessing the QLPS Network page:

`138d`

Where:

d = QLPS network associated with the target GQPH QPipe

Result:

Check the results and take one of the following actions:

- If **neither** of the QLPSes is **ACT**, then at least one of the QLPSes must be recovered. EXIT this procedure and refer to the Corrective Maintenance procedure, *Clear Quad Link Packet Switch Problems*, 235-105-220, to repair the QLPS. After the QLPS is recovered, RE-ENTER this procedure at Step 1.
- If only one QLPS is **ACT** in network **d** but one or more child **QTMSLNKs** is **OOS**, then request a detailed status of the GQPH QPipe:
  `OP:STATUS,GQPHPIPE,QPIPE=a-b-c-d-e,DETAIL;`
  Where:
  a = Global switch module number
  b = Packet switch unit 2 number
  c = PSU2 shelf number
  d = GQPH channel group number
  e = QLPS network number
  If the QTMSLNK port in the output matches an **OOS QTMSLNK** on the target **ACT QLPS**, then EXIT this procedure and refer to the Corrective Maintenance procedure, *Correct QTMSLNK Problems*, 235-105-220, to repair the QTMSLNK. After the link is recovered, RE-ENTER this procedure at Step 1.
- If one QLPS is **ACT** in network **d** and no child **QTMSLNK** is **OOS**, then request a detailed status of the GQPH QPIPE:
  `OP:STATUS,GQPHPIPE,QPIPE=a-b-c-d-e,DETAIL;`
  Where:
  a = Global switch module number
  b = Packet switch unit 2 number
  c = PSU2 shelf number
  d = GQPH channel group number
  e = QLPS network number
  The corresponding primary network link interface (NLI) serving the **ACT QLPS** on QLPS network **d** is specified in the output. Only a single digit **j** (0 or 1) is specified and actually **NLI a-j-k**, where **k** is the **ONTC** with the active QLPS, is serving the target GQPH QPipe.
  Access the NLI Summary page:
  `1200,X`

Locate the NLI of interest and display the corresponding DLI/NLI/TMSLNK SET page:

`1201,Y`

Determine if the NLI itself or the first associated TMSLNK is **OOS**. If one is found, then EXIT this procedure now and refer to the Corrective Maintenance procedure, *Clear Network Link Interface Problems*, 235-105-220, to repair the NLI or TMSLNK. After the parent TMSLNK hardware is recovered, RE-ENTER this procedure at Step 1 to verify new GQPH QPipe status and take appropriate action.

- If no problem with parent QTMSLNK/TMSLNK has been identified by now, then the OOSF-CM GQPH QPipe status must have been recovered by independent actions. RETURN to Step 1 to verify the new GQPH QPipe status and take the appropriate action.

---

4 An **OOSF-PH** GQPH QPipe has been removed because the parent GQPH channel group is unassigned.

Obtain the status of the PH33:

`118X,Y,ZZZ`

Where:

X = Packet switch unit 2 shelf number

Y = PSU2 number

ZZZ = Global switch module number

The associated GQPH channel group **d** should be unassigned and there should be at least one **OOS PH33** on the shelf. If **no** PH has an **OOSF** status displayed, obtain a list of the OOS PH33s and recover at least one of them using the SIP for Packet Trunking Corrective Maintenance procedure, *Clear Protocol Handler Problems*, 235-200-118. If **all** equipped PHs are **OOSF**, then the parent PSUCOMs are duplex failed. Refer to the System Recovery [5E14 and Later Software Releases] procedure, *PSU Duplex Failure*, 235-105-250, to recover one of the PSUCOMs.

After the PH33/PSUCOM is recovered, RE-ENTER this procedure at Step 1.

5 An **OOS-GQPHLB** GQPH QPIPE has been removed because of a local loopback failure in the PH or a failure to activate the appropriate network processor (NP) channel. The parent physical PH is faulty, operating in “degraded” mode if no STBY PH33 is on the same shelf. This status is typically a transitional state but you may want to verify the status:

`OP:STATUS,GQPHPIPE,QPIPE=a-b-c-d-e;`

Where:

a = Global switch module number

b = Packet switch unit 2 number

c = PSU2 shelf number

d = GQPH channel group number

e = QLPS network number

If it still shows **OOS-GQPHLB**, obtain the status of the PH33:

`118X,Y,ZZZ`

Where:

X = Packet switch unit 2 shelf number

Y = PSU2 number

ZZZ = Global switch module number

The associated GQPH channel group **d** should be assigned to a **DEGR** PH33 and there should be at least one **OOS PH33** on the shelf. Obtain a list of **OOS PH33s** and recover at least one of them using the SIP for Packet Trunking Corrective Maintenance procedure, *Clear Protocol Handler Problems*, 235-200-118.

If there is a spare or STBY PH33 on the shelf, it might help to switch the GQPH channel group with the failed pipes to the STBY PH33.

After the PH33 is recovered, RE-ENTER this procedure at Step 1 because there may be other problems that inhibit the target GQPH QPipe from being successfully restored.

6 An **OOS-QLPSLB** GQPH QPipe cannot successfully loop back a test frame to the QLPS. The TSI-QPH nailed-up path is established and the NP channel is activated, and an undetected failure is presumed to be in the communications module (CM) complex. The parent PH will periodically re-execute the QLPS loopback test and the GQPH QPipe may be automatically recovered. Also, the SMP will schedule a full restoral periodically.

An **OOS-LVL1-FRAME** GQPH QPipe indicates that framing errors have been detected at the NP channel. The nailed-up TSI-QPH path is torn down and the NP channel is deactivated, and an undetected failure is presumed to be in the CM complex. The parent SMP will periodically attempt a complete restoral and the GQPH QPipe may be automatically recovered.

7 It is possible that undetected faults in parent QTMSLNK/TMSLNK hardware may be the reason that a GQPH QPipe is in the **OOS-QLPSLB** or **OOS-LVL1-FRAME** state.

The first step in the analysis is to determine the state of parent hardware to be recovered.

Access the QLPS Network status page:

```
138d
```

Where:

d = QLPS network associated with the target GQPH QPipe. Record the QLPS in the target network.

If the mate QLPS in the target QLPS network is **OOS**, which may be due to parent ONTCCOM outage, then recover that QLPS. Refer to the Corrective Maintenance procedure, *Clear Quad Link Packet Switch Problems,* 235-105-220, to repair the QLPS.

If any QTMSLNKs are OOS, then refer to the Corrective Maintenance procedure, *Correct QTMSLNK Problems,* 235-105-220, to repair the identified QTMSLNKs.

Obtain a detailed status of the GQPH QPipes

```
OP:STATUS,GQPHQPIPE,QPIPE=a-b-c-d-e,DETAIL;
```

Where:

a = Global switch module number

b = Packet switch unit 2 number

c = PSU2 shelf number

d = GQPH channel group number

e = QLPS network number

The corresponding primary NLI serving the **ACT QLPS** on QLPS network **d** is specified in the output. Only a single digit **j** (0 or 1) is specified and actually **NLI a-j-k**, where **k** is the **ONTC** side, could serve GQPH Qpipe on the target QLPS network.

Access the NLI Summary page:

`1200,X`

Locate the NLI of interest and display the corresponding DLI/NLI/TMSLNK SET page:

`1201,Y`

Determine if the NLI itself or the first associated TMSLNK is **OOS**. If one is found, then refer to the Corrective Maintenance procedure, *Clear Network Link Interface Problems,* 235-105-220, to repair the NLI or TMSLNK.

After all QLPSs, QTMSLNKs and GSM NLIs on the target QLPS network have been removed, did the associated QLPSs switch?

- If **yes**, check the GQPH QPIPE status
  `OP:STATUS,GQPHQPIPE,QPIPE=a-b-c-d-e;`
  Where:
  a = Global switch module number
  b = Packet switch unit 2 number
  c = PSU2 shelf number
  d = GQPH channel group number
  e = QLPS network number
  If the resultant state is **OOS-QLPSLB** or **OOS-LVL1-FRAME**, CONTINUE this procedure for further recovery.
  Otherwise, RETURN to Step 1 because the status of multiple GQPH QPIPEs may have been affected by hardware reconfigurations.
- If **no**, attempt to conditionally switch QLPSs in the target network
  `SW:QLPS=0-d;`
  If the switch is successful, then determine the status
  `OP:STATUS,GQPHQPIPE,QPIPE=a-b-c-d-e;`
  Where:
  a = Global switch module number

b = Packet switch unit 2 number
c = PSU2 shelf number
d = GQPH channel group number
e = QLPS network number
If the resultant state is **OOS-QLPSLB** or **OOS-LVL1-FRAME**, CONTINUE this procedure for further recovery.
Otherwise, RETURN to Step 1 because the status of multiple GQPH QPipes may have been affected by hardware reconfigurations.

8 In a further attempt to isolate undetected problems in parent CM hardware, attempt to conditionally restore each QLPS in target network **d**

```
RST:QLPS=a-d;
```

Where:

a = the ONTC side, noting the request may be denied because of higher level problems.

If the conditional restore fails, then refer to the Corrective Maintenance procedure, *Clear Quad Link Packet Switch Problems*, 235-105-220, to repair/restore the QLPS.

If either QLPS was repaired or restored, recheck target GQPH QPipe status:

```
OP:STATUS,GQPHQPIPE,QPIPE=a-b-c-d-e;
```

Where:

a = Global switch module number

b = Packet switch unit 2 number

c = PSU2 shelf number

d = GQPH channel group number

e = QLPS network number

If the resultant state is **NOT OOS-QLPSLB** or **OOS-LVL1-FRAME**, then RETURN to Step 1 because the status of multiple GQPH QPipes may have been affected by hardware reconfigurations.

Otherwise, attempt to conditionally switch QLPSs in the target network:

```
SW:QLPS=0-d;
```

If the switch is successful, obtain the QPIPE status:

```
OP:STATUS,GQPHQPIPE,QPIPE=a-b-c-d-e;
```

Where:

a = Global switch module number

b = Packet switch unit 2 number

c = PSU2 shelf number

d = GQPH channel group number

e = QLPS network number

If the resultant state is **OOS-QLPSLB** or **OOS-LVL1-FRAME**, CONTINUE this procedure for further recovery.

Otherwise, RETURN to Step 1 because the status of multiple GQPH QPipes may have been affected by hardware reconfigurations.

9 In a further attempt to isolate undetected problems in parent CM hardware, attempt to conditionally restore each GSM primary NLI in target network **d**.

```
OP:STATUS,GQPHQPIPE,QPIPE=a-b-c-d-e,DETAIL;
```

Where:

a = Global switch module number

b = Packet switch unit 2 number

c = PSU2 shelf number

d = GQPH channel group number

e = QLPS network number

The corresponding primary NLI serving the ACT QLPS on QLPS network **d** is specified in the output. Only a single digit **j** (0 or 1) is specified and actually **NLI a-j-k**, where **k** is the **ONTC** side could serve GQPH QPipe on the target QLPS network.

```
RST:NLI=a-j-k;
```

If the conditional restore fails, then refer to the Corrective Maintenance procedure, *Clear Network Link Interface Problems,* 235-105-220, to repair the NLI.

If either NLI was repaired or restored, recheck the target GQPH QPipe status:

`OP:STATUS,GQPHQPIPE,QPIPE=a-b-c-d-e;`

Where:

a = Global switch module number

b = Packet switch unit 2 number

c = PSU2 shelf number

d = GQPH channel group number

e = QLPS network number

If the resultant state is **NOT OOS-QLPSLB** or **OOS-LVL1-FRAME**, then RETURN to Step 1 because the status of multiple GQPH QPipes may have been affected by hardware reconfigurations.

Otherwise, attempt to conditionally switch QLPSs in the target network:

`SW:QLPS=0-d;`

If the switch is successful, obtain of the QPipe

`OP:STATUS,GQPHQPIPE,QPIPE=a-b-c-d-e;`

Where:

a = Global switch module number

b = Packet switch unit 2 number

c = PSU2 shelf number

d = GQPH channel group number

e = QLPS network number

If the resultant state is **NOT OOS-QLPSLB** or **OOS-LVL1-FRAME**, CONTINUE this procedure for further recovery.

Otherwise, RETURN to Step 1 because the status of multiple GQPH QPipes may have been affected by hardware reconfigurations.

10 An **OOS-LVL1-PATH** GQPH QPipe has failed a restoral because the nailed-up TSI-QPH path could not be established. An assert may be produced, pointing to an ODD problem, or there may be a subtle error in timeslot assignment. This step may also be reached if previous

OOS-QLPSLB or **OOS-LVL1-FRAME** GQPH QPipe recovery has failed.

Access the MCTSI status page:

`1800,a`

Result:

If either MCTSI is forced `ACT`, clear the force using the **422** menu selection. If no force condition is present but an OOS/UNAV MCTSI is displayed, restore the MCTSI:

`RST:MCTSI,SM=a;`

If the prediagnostic phase of the restore fails, then refer to the Corrective Maintenance procedure, *Clear Diagnostic Failure in Hardware Units/Circuits,* 235-105-220, to repair the unit.

If the status of the MCTSI is **ACT/STBY**, issue the following command

`SM:MCTSI,SM=a;`

This will bring in new resources, possibly resolving undetected MCTSI problems and the MCTSI switch will trigger automatic recovery of the target GQPH QPipe.

Verify that status of the target GQPH QPipe:

`OP:STATUS,GQPHQPIPE,QPIPE=a-b-c-d-e;`

Where:

a = Global switch module number

b = Packet switch unit 2 number

c = PSU2 shelf number

d = GQPH channel group number

e = QLPS network number

If the resultant state is **OOS-QLPSLB, OOS-LVL1-FRAME** or **OOS-LVL1-PATH**, CONTINUE this procedure for further recovery.

Otherwise, RETURN to Step 1 because the status of multiple GQPH QPipes may have been affected by MCTSI hardware reconfigurations.

11 The PH33 supporting the GQPH channel group may encounter corruption that can cause erroneous **OOS-QLPSLB, OOS-LVL1-FRAME** or **OOS-LVL1-PATH** GQPH QPipe status. This rare occurrence may be resolved by removing and restoring PH33 supporting the target GQPH channel group.

Obtain the status of the PH33:

```
118X,Y,ZZZ
```

Where:

X = Packet switch unit 2 shelf number

Y = PSU2 number

ZZZ = Global switch module number

Result:

Identify all `STBY` PH33s on the target shelf. Remove those `STBY` PH33s using **2XX, UCL** menu selection. Immediately restore all OOS PH33s that were removed using **3XX,UCL** menu selection.

If the resultant state is **OOS-QLPSLB, OOS-LVL1-FRAME** or **OOS-LVL1-PATH**, then no further recovery is possible. Gather Supplementary error reports, if any, on the ROP with headers. Refer to the SIP for Packet Trunking Corrective Maintenance procedure, *Analysis of General Message Transport Error Reports*, 235-200-118, for the target GQPH QPipe as well as any associated asserts/audits and SEEK TECHNICAL ASSISTANCE according to local procedures.

Otherwise, RETURN to Step 1 because the status of mate GQPH QPipe on the target channel group may also be affected by the PH hardware reconfiguration.

12 An **INIT** GQPH QPIPE is currently being initialized or parent SMP will periodically attempt to complete restoral and the GQPH QPipe may be automatically recovered. In any of these cases, it is wise to manually remove a GQPH QPipe unconditionally:

```
RMV:GQPHPIPE,QPIPE=a-b-c-d-e,UCL;
```

Where:

a = Global switch module number

b = Packet switch unit 2 number

c = PSU2 shelf number

d = GQPH channel group number

e = QLPS network number

Then manually restore the GQPH QPipe:

```
RST:GQPHQPIPE=a-b-c-d-e;
```

Result:

Based on the result, take one of the following actions:

- If the restore is successful, check for additional known OOS GQPH QPipes to recover.
- If the restore failed and the resulting state is still **INIT**, there is no further recovery possible. Gather any Supplementary error reports on the ROP with headers. Refer to the SIP for Packet Trunking Corrective Maintenance procedure, *Analysis of General Message Transport Error Reports*, 235-200-118, for the target GQPH QPipe as well as any associated asserts/audits and SEEK TECHNICAL ASSISTANCE according to local procedures. Otherwise, RETURN to Step 1 to re-analyze the GQPH QPipe based on the new resultant state.

13 An **UNKNOWN** GQPH QPipe is most likely in a transient state and the state should stabilize shortly.

Re-verify the status of the target GQPH QPipe:

```
OP:STATUS,GQPHPIPE,QPIPE=a-b-c-d-e;
```

Where:

a = Global switch module number

b = Packet switch unit 2 number

c = PSU2 shelf number

d = GQPH channel group number

e = QLPS network number

If the resultant state is again **UNKNOWN**, then there is no further recovery possible. Gather any associated asserts/audits and SEEK TECHNICAL ASSISTANCE according to local procedures.

Otherwise, RETURN to Step 1 to re-analyze the GQPH QPipe based on the new resultant state.

14 Stop. You have completed this procedure.

END OF STEPS

Analyze General Message Transport (GMT) Error Reports

Purpose

This procedure addresses failure reports of internal message transport that is detected by recovery software. Autonomous reports provide enough information to identify the GSM/NGSM combinations with GMT problems or possibly GQPH QPipes or GQPH Links (GQPHLNKs) causing these problems.

Note: Analyze General Message Transport (GMT) Error Reports does not apply to DRM/VCDX.

When to use

Procedural steps point to appropriate problem clearing based on GMT mechanism/error type identified. When multiple reports are generated, indicating different levels of GMT disruptions have occurred, resolution of the problems should follow the severity order.

Related information

Refer to the *Input Messages Manual*, 235-600-700, and the *Output Messages Manual*, 235-600-750, for additional information.

Admonishments

These autonomous messages indicate that a GMT problem has occurred some time in the past, so the problems may have already been addressed by automatic recovery strategies, or some of the dependent hardware resources may have been recovered or repaired independently before intervention. Reports may also be a result of valid maintenance personnel actions, even though signaling operation is severely impacted.

Before you begin

Required Tools

- Use of the Master Control Center (MCC) or Supplementary Trunk Line Workstation (STLWS).

Required Material

- The switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking feature software.

Required Information

- None.

Required Conditions

- None required.

Procedure

1 **NOTE:** Resolve the highest level problem identified in the steps below. Repeat this process until all identified problems have been reconciled.

2 The most severe message disruption occurs when the following message is generated:

REPT GENERAL MESSAGE TRANSPORT CONNECTIVITY CHANGE

GSM=a SERV=b ALL GMT CONNECTIVITY LOST

TIMESTAMP:c

Where:

a = Global switch module number

b = SIP (Service Type)

c = Initial GMT loss/recovery timestamp

This message reports loss of GMT connectivity for all provisioned NGSM-2000s which implies that all SIP signaling activity supported by the PSU2 platform on the GSM has ceased. The original report will probably be CRITICALLY alarmed and will be repeated every 15 minutes.

This usually indicates that all GQPH QPipes are OOS on this GSM. Refer to the SIP for Packet Trunking Corrective Maintenance procedure, *Resolve GQPH QPipe Problems*, 235-200-118, to recover the identified GQPH QPipes.

3 A similar but less severe message disruption occurs when the following message is generated:

REPT GENERAL MESSAGE TRANSPORT CONNECTIVITY CHANGE

GSM=a SERV=b c

TIMESTAMP:d

NGSMs AFFECTED:e

Where:

a = Global switch module number

b = SIP (Service type)

c = Interconnectivity status

- ACCESSIBLE = NGSMs become accessible.
- INACCESSIBLE = NGSMs become inaccessible.

d = Initial GMT loss/recovery timestamp

e = NGSMs impacted

This message reports that one or more NGSM-2000s have no GMT connectivity to a particular GSM. This usually indicates that the GSM is initializing.

Note: The NGSM-2000s in the message and list all the GQPH QPipes for GSM:

```
OP:STATUS,GQPHPIPE,GSM=a;
```

Where:

a = Global switch module number

Result:

Note: The GQPH Qlinks for the particular QPipes and obtain the status of the Qlinks for every NGSM-2000s:

```
OP:STATUS,GQPHLNK,GQPH=a-b-c-d;
```

Where:

a = Global module switch number

b = Packet switch unit 2 number

c = PSU2 shelf number

d = GQPH channel group number

Refer to the *Output Messages Manual*, 235-600-750, and take appropriate actions.

4 The following message indicates that one or more GQPH QPipes have undergone recovery:

REPT GQPHPIPE RECOVERY GSM=x SERV=y

Where:

x = Global switch module number

y = SIP (Service Type)

Examine the **FINAL STATUS** entry of the GQPH QPipe listed and take appropriate action.

- If the final status is **ACT** and the **RECOVERY** is not **TRANSIENT OUTAGE**, then the GQPH QPipe has automatically recovered by reconfiguring parent hardware. Independent craft personnel actions will be take to repair any faulty hardware but no direct action need to be taken because GMT has not been permanently impacted.
- If the final status is **ACT** and the **RECOVERY** is **TRANSIENT OUTAGE**, then the GQPH QPipe has automatically recovered but the reason for the outage is unknown.
  No direct action need be taken because GMT has not been permanently impacted, but the reason for outage should be investigated. Gather supplementary error reports with headers and SEEK TECHNICAL ASSISTANCE according to local procedures.
- If the final status is **not ACT**, then one or more GQPH QPipes on a particular GSM has been removed. Record the inactive GQPH QPipes and refer to the SIP for Packet Trunking Corrective Maintenance procedure, *Resolve GQPH QPipe Problems,* 235-200-118, to recover the identified GQPH QPipes.

---

5 If the supplementary unalarmed error report with headers generated is:

REPT ERROR DATA GQPHPIPE=a-b-c-d-e

Where:

a = Global switch module number

b = Packet switch unit 2 number

c = PSU2 shelf

d = GQPH channel group

e = QLPS network

Refer to the SIP for Packet Trunking Corrective Maintenance procedure, *Resolve GQPH QPipe Problems,* 235-200-118, to possibly recover the GQPH QPipes. However, the failure of identified GQPH QPipes should be investigated. SEEK TECHNICAL ASSISTANCE according to local procedures.

6 If the supplementary unalarmed error report with headers generated is:

REPT ERROR DATA GQPHLNK=a-b-c-d-e-f

Where:

a = Global switch module number

b = Packet switch unit 2 number

c = PSU2 shelf

d = GQPH channel group

e = QLPS network

f = NGSM-2000 number

This report indicates that the identified **GQPHLNK** has been removed by fault recovery, but both the GQPH QPipes are MH QPipes are still in service, implying subtle software problems.

Gather supplementary reports and SEEK TECHNICAL ASSISTANCE according to local procedures.

7 If the message generated is:

OP OVRLD or **REPT OVERLOAD HISTORY**

Overloads possibly impacted by GMT traffic are resource types **MHRT** (MH real-time overload), **MHPIPE** (MH QPipe occupancy overload) and **GSMQPH** (GQPH real-time or resource overload).

8 Stop. You have completed this procedure.

END OF STEPS

Resolve SCTP Association Problems

Purpose

This procedure identifies methods for resolving associations with `CLOSED` status.

When to use

Use this procedure to restore a SCTP association when an association transitions from `ESTABLISHED` or `DEGRADED` to `CLOSED` or a status request indicates `CLOSED`.

Related information

Refer to the *Input Messages Manual*, 235-600-700, and the *Output Messages Manual*, 235-600-750, for additional information.

Admonishments

Unconditionally removing an association can be service-affecting causing the shutdown of traffic.

Before you begin

Required Tools

- Use of the Master Control Center (MCC) or Supplementary Trunk Line Workstation (STLWS).

Required Material

- The switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking feature software.

Required Information

- Association number to be recovered is known.

Required Conditions

- Association status is `CLOSED`.

Procedure

1 If the status is `CLOSED MAN`, restore the association:

```
RST:SCTP,ASSOCIATION=a;
```

Where:

a = association number

Result:

If restoral fails, then refer to the SIP for Packet Trunking Corrective Maintenance procedure, *Restore SCTP Association*, 235-200-118.

2 If the status is `CLOSED ENDPT`, then the problem is with the SCTP near endpoint of the association. Refer to the SIP for Packet Trunking Corrective Maintenance procedure, *Resolve SCTP Near Endpoint*, 235-200-118.

3 If the status is `CLOSED ERROR`, first remove the association:

`RMV:SCTP,ASSOCIATION=a[,UCL];`

Where:

a = association number

Result:

Observe `COMPLETED` output message.

4 After the association was successfully removed, then restore it:

`RST:SCTP,ASSOCIATION=a;`

Where:

a = association number

Result:

If the association is still `CLOSED ERROR`, the problem is most likely with the far end of the association or IP network in between or outside of the *5ESS*® switch.

5 Stop. You have completed this procedure.

END OF STEPS

Remove SCTP Association

Purpose This procedure requests the removal of an SCTP association.

When to use Use this procedure when the removal of an SCTP association is required.

Related information Refer to the *Input Messages Manual*, 235-600-700, and the *Output Messages Manual*, 235-600-750, for additional information.

Admonishments Unconditionally removing an association can be service-affecting causing the shutdown of traffic.

Before you begin

Required Tools

- Use of the Master Control Center (MCC) or Supplementary Trunk Line Workstation (STLWS).

Required Material

- The switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking feature software.

Required Information

- Association number to be removed is known.

Required Conditions

- Association status is not already `CLOSED`.

Procedure

1 Request a removal of a specific association:

`RMV:SCTP,ASSOC=b,[UCL];`

Where:

b = association number

Result:

Reports the result of removing a specific association from service.

2 If the result is `NG - INVALID ASSOCIATION ID`, then check the association number on RC/V 33.22. For all others, refer to the *Output Messages Manual*, 235-600-750, for additional information.

3 Stop. You have completed this procedure.

END OF STEPS

Restore SCTP Association

Purpose

This procedure requests the restoral of an SCTP association.

When to use

Use this procedure when the restoral of an SCTP association is required.

Related information

Refer to the *Input Messages Manual*, 235-600-700, and the *Output Messages Manual*, 235-600-750, for additional information.

Admonishments

None.

Before you begin

Required Tools

- Use of the Master Control Center (MCC) or Supplementary Trunk Line Workstation (STLWS).

Required Material

- The switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking feature software.

Required Information

- Association number to be restored is known.

Required Conditions

- Association status is `CLOSED`.

Procedure

1 Request a restoral of a specific association :

```
RST:SCTP,ASSOC=b;
```

Where:

b = association number

Result:

Reports the result of restoring a specific association to service.

2 If the result is `NG - INVALID ASSOCIATION ID`, then check the association number on RC/V 33.22. For all others, refer to the *Output Messages Manual*, 235-600-750, for additional information.

3 Stop. You have completed this procedure.

END OF STEPS

☐

Resolve SCTP Near endpoint Problems

Purpose

This procedure identifies methods for resolving SCTP near endpoint with NOSERV or OOS status.

When to use

Use this procedure when an SCTP near endpoint transitions from

- `INSERV` to `PARTSERV`,
- `INSERV` to `NOSERV` or `OOS`, or
- `PARTSERV` to `NOSERV` or `OOS`

Related information Refer to the *Input Messages Manual*, 235-600-700, and the *Output Messages Manual*, 235-600-750, for additional information.

Admonishments Unconditionally removing an endpoint can be service-affecting causing the shutdown of traffic.

Before you begin

Required Tools

- Use of the Master Control Center (MCC) or Supplementary Trunk Line Workstation (STLWS).

Required Material

- The switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking feature software.

Required Information

- Near endpoint name to be recovered is known.

Required Conditions

- Near endpoint status is `NOSERV` or `OOS` due to a status request or the autonomous **REPT SCTP NEAREPT**.

Procedure

1 Obtain a detail status report of endpoint, if the status is not already known:

`OP:STATUS,SCTP,NEAREPT=a,DETAIL;`

2 If the status is `OOS MAN`, restore the endpoint:

`RST:SCTP,NEAREPT=a;`

Where:

a = near endpoint name

Result:

If the endpoint remains OOS, then refer to the SIP for Packet Trunking Corrective Maintenance procedure, *Restore SCTP Near Endpoint*, 235-200-118.

3 If the status is OOS UNAVL, the problem is that there is no SERVING SIP PH in the processor group that supports the endpoint, or that the SERVING SIP PH has a PING DOWN status.

For each GSM, check the status of all processor groups:

`OP:STATUS,SERV,PCRGRP=a-1&&24;`

Where:

x = Global switch module number

Result:

If any of the processor groups do not have a SIP PH designated as SERVING:

- If the PH(s) in the processor group are OOS, attempt to restore at least one of the PHs:
  `RST:PSUPH=a-b-c-d;`
  Where:
  a = Global switch module number
  b = Packet switch unit 2 number
  c = PSU2 shelf number
  d = PHE2 number
- If the Ethernet link is OOS, then determine the cause of the Ethernet link going down and repair it, if possible.

If the SIP PH is designated as SERVING but has a PING status DOWN, then problem is most likely with the adjacent IP router external to the *5ESS*® switch.

4 If the status is NOSERV, first remove the near endpoint unconditionally:
`RMV:SCTP,NEAREPT=a,UCL;`

Where:

a = endpoint name

Result:

The endpoint is removed.

5 After the near endpoint was successfully removed, then restore it:
`RST:SCTP,NEAREPT=a;`

Where:

a = endpoint name

Result:

If the restore fails, then refer to the SIP for Packet Trunking Corrective Maintenance procedure, *Restore SCTP Near Endpoint*, 235-200-118.

6 Stop. You have completed this procedure.

END OF STEPS

Remove SCTP Near Endpoint

Purpose

This procedure requests the removal of an SCTP Near Endpoint.

When to use

Use this procedure when the removal of an SCTP Near Endpoint is required.

Related information

Refer to the *Input Messages Manual,* 235-600-700, and the *Output Messages Manual,* 235-600-750,for additional information.

Admonishments

Unconditionally removing an endpoint can be service-affecting causing the shutdown of traffic.

Before you begin

Required Tools

- Use of the Master Control Center (MCC) or Supplementary Trunk Line Workstation (STLWS).

Required Material

- The switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking feature software.

Required Information

- Near endpoint name to be removed is known.

Required Conditions

- Endpoint status is not already NOSERV or OOS.

Procedure

1 Request a removal of a specific near endpoint:

```
RMV:SCTP,NEAREPT=a,[UCL];
```

Where:

a = near endpoint name

Result:

Reports the result of removing a specific near endpoint from service.

2 Check the result and take appropriate action:

- If the result is `NG - INVALID NEAR ENDPOINT NAME` then either dump all the provisioned endpoints or examine the associated office record.
  Refer to *Translation Guide (TG-5)*, 235-080-100, for obtaining office record numbers.
- For all others, refer to the *Output Messages Manual*, 235-600-750, for additional information.

3 Stop. You have completed this procedure.

END OF STEPS

□

Restore SCTP Near Endpoint

Purpose This procedure requests the restoral of an SCTP near endpoint.

When to use Use this procedure when the restoral of an SCTP near endpoint is required.

Related information Refer to the *Input Messages Manual*, 235-600-700, and the *Output Messages Manual*, 235-600-750, for additional information.

Admonishments None

Before you begin

Required Tools

- Use of the Master Control Center (MCC) or Supplementary Trunk Line Workstation (STLWS).

Required Material

- The switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking feature software.

Required Information

- None required.

Required Conditions

- Endpoint is `NOSERV` or `OOS`.

Procedure

1 Request a restoral of a specific near endpoint :

```
RST:SCTP,NEAREPT=a;
```

Where:

a = near endpoint name

Result:

Reports the result of restoring a specific near endpoint to service.

2 Check the results and take appropriate action:

- If the result is `NG - INVALID NEAR ENDPOINT NAME`, then either dump all the provisioned endpoints or examine the associated office record.
  Refer to *Translation Guide (TG-5)*, 235-080-100, for obtaining office record numbers.
- For all others, refer to the *Output Messages Manual*, 235-600-750, for additional information.

3 Stop. You have completed this procedure.

END OF STEPS

□

Analyze SCTP Near Endpoint Change Report

Purpose

This procedure provides an analysis of an SCTP near endpoint change report when an SCTP endpoint autonomously transitions from one state to another.

When to use

Use this procedure when an SCTP endpoint change report is generated.

Related information

Refer to the *Input Messages Manual*, 235-600-700, and the *Output Messages Manual*, 235-600-750, for additional information.

Admonishments

State transitions associated with manual removals or restorals of an endpoint are not included.

Before you begin

Required Tools

- Use of the Master Control Center (MCC) or Supplementary Trunk Line Workstation (STLWS).

Required Material

- The switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking feature software.

Required Information

- None required.

Required Conditions

- None required.

Procedure

1 Analyze the REPT SCTP ENDPOINT change report:

REPT SCTP NEAREPT=a

PCRGRP=c-d-e PSUPH=c-d-f-g SHELF-CHGRP=f-h

PREV-STATE=i NEW STATE=j

Where:

a = Near endpoint name

c = Switching module number

d = Packet switch unit 2 number

e = Processor group number

f = Packet switch unit 2 shelf number

g = PHE2 number

h = Channel group number

i = Status of an endpoint or association before the autonomous transition

j = Status of endpoint or association after the autonomous transition

2 Check the result and take appropriate action:

- If the REPT SCTP ENDPOINT change report is generated with preceding asterisks, then this is a major alarm indicating a near endpoint transitioned to `NOSERV`. Refer to the SIP for Packet Trunking Corrective Maintenance procedure, *Resolve SCTP Near Endpoint Problems*, 235-200-118.
- If the REPT SCTP ENDPOINT change report is generated without preceding asterisks, then make note of the change and determine whether further actions are required.

3 Stop. You have completed this procedure.

END OF STEPS

Analyze SCTP Association Change Report

Purpose

This procedure provides an analysis of an SCTP association change report when an SCTP association autonomously transitions from one state to another.

When to use

Use this procedure to analyze an SCTP association change report.

Related information

Refer to the *Input Messages Manual*, 235-600-700, and the *Output Messages Manual*, 235-600-750, for additional information.

Admonishments

State transitions associated with manual removals or restorals of an association are not included.

Before you begin

Required Tools

- Use of the Master Control Center (MCC) or Supplementary Trunk Line Workstation (STLWS).

Required Material

- The switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking feature software.

Required Information

- None required.

Required Conditions

- None required.

Procedure

1 Analyze the REPT SCTP ASSOCIATION change report generated:

REPT SCTP ASSOC ID=b

NEAR ENDPT=a FAR ENDPOINT=k

PCRGRP=c-d-e PSUPH=c-d-f-g SHELF-CHGRP=f-h

PREV-STATE=i NEW STATE=j

Where:

a = Near endpoint name

b = Association number

c = Switching module number

d = Packet switch unit 2 number

e = Processor group number

f = Packet switch unit 2 shelf number

g = PHE2 number

h = Channel group number

i = Status of endpoint or association before the autonomous transition

j = Status of endpoint or association after the autonomous transition

k = Far endpoint name

2 Check the result and take appropriate action:

- If the REPT SCTP ASSOCIATION change report is generated with preceding asterisks, then this is a major alarm indicating that an association transitioned to CLOSED ERROR. Refer to the SIP for Packet Trunking Corrective Maintenance procedure, *Resolve SCTP Association Problems*, 235-200-118.
- If the REPT SCTP ASSOCIATION change report is generated without preceding asterisks, then make note of the change and determine whether further actions are required.

3 Stop. You have completed this procedure.

END OF STEPS

Request Manual Service Selection Switch

Purpose

This procedure requests that the service selection of a processor group be switched.

When to use

Use this procedure when a manual service selection switch of a processor group is required.

Related information

Refer to the *Input Messages Manual*, 235-600-700, and the *Output Messages Manual*, 235-600-750, for additional information.

Admonishments

A manual switch is not permitted with a processor group of one.

Before you begin

Required Tools

- Use of Master Control Center (MCC) or Supplementary Trunk Line Workstation (STLWS).

Required Material

- The switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking feature software.

Required Information

- Refer to Office Record 5880 for provisioned GSMs.
- Refer to Office Record 5883 for provisioned processor groups.

Required Conditions

- The processor group is equipped with two processors.
- One processor in the processor group is SERVING and the other is NON-SERVING.

Procedure

1 Request a manual service selection switch for a specific PSU PH:

```
SW:SERV,PSUPH=a-c-d-e;
```

Where:

a = Global switch module number

c = Packet switch unit 2 number

d = PSU2 shelf number

e = PH position number

Result:

Generates the service selection switch report for a specific PSU2.

2 Check results and take appropriate action:

- If the switch `FAILED`, then probably a software error occurred. SEEK TECHNICAL ASSISTANCE according to local procedures for the affected PH.
- If the switch is `DENIED`, then a wrong identifier was entered. Retry with the correct identifier.
- If the response is `ELNK DOWN`, then determine the cause of the Ethernet link going down and repair it, if possible.
- If the response is `PING FAILURE`, then the problem is most likely with the adjacent IP router external to the *5ESS*® switch.

3 Stop. You have completed this procedure.

END OF STEPS

Analyze Service Selection Change Report

Purpose

This procedure provides an analysis of the service selection change report which is a result of an automatic or manual action of a processor group changing its service selection.

When to use

Use this procedure when a service selection change report is generated.

Related information

Refer to the *Input Messages Manual,* 235-600-700, and the *Output Messages Manual,* 235-600-750, for additional information.

Admonishments

None

Before you begin

Required Tools

- Use of the Master Control Center (MCC) or Supplementary Trunk Line Workstation (STLWS).

Required Material

- The switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking feature software.

Required Information

- None required.

Required Conditions

- None required.

Procedure

1 Analyze the `REPT SERV PCRGRP` change report generated:

REPT SERV PCRGRP=a-b

PSUPH=a-c-d-e f g

Where:

a = Global switch module number

b = Processor group number

c = Packet switch unit 2 number

d = PSU2 shelf number

e = PH position number

f = Service selection state for the processor identified

g = Reason for the service selection change

2 If the `REPT SERV PCRGRP` change report is generated without errors, then make note of the change and EXIT this procedure. Otherwise, CONTINUE with this procedure.

3 Check the reason for the change and take appropriate action:

- If the reason for the service selection change is AUTO RMV, then refer to the SIP for Packet Trunking Corrective Maintenance procedure, *Resolve PH Problems*, 235-200-118.
- If the reason for the service selection change is `ELNK DOWN`, then determine the cause of the Ethernet link going down, e.g. loose or faulty cable, and repair it, if possible. A corresponding `REPT ELNK` report should have been generated.
  However, it is possible that the LLE2 paddleboard is faulty. Refer to the SIP for Packet Trunking Corrective Maintenance procedure, *Resolve LLE2 Paddleboard problems*, 235-200-118.

If the service selection change report is generated with preceding asterisks, then this is a major alarm indicating that a SIP PH transitioned from serving to unavailable.

This alarmed report usually indicates that the PHE2 is OOS, the corresponding LLE2 paddleboard is faulty or the Ethernet link is down. Refer to the SIP for Packet Trunking Corrective Maintenance procedure, *Resolve PH Problems*, 235-200-118.

If the service selection change report is generated with a preceding “*C”, then this is a critical alarm indicating that a SIP PH transitioned from serving to unavailable and the other SIP PH in the processor group is unavailable or unequipped.

This alarmed report usually indicates that the PHE2 is `OOS`, the corresponding LLE2 paddleboard is faulty or the Ethernet link is down. Refer to the SIP for Packet Trunking Corrective Maintenance procedure, *Resolve PH problems*, 235-200-118.

4 Stop. You have completed this procedure.

END OF STEPS

Dump IP Routing Table

Purpose This procedure requests a dump of IP routing tables.

When to use Use to troubleshoot layer 3 network connectivity problems between the *5ESS*® switch and the external network. In particular, verify connectivity between a SIP PH and an adjacent router.

Related information Refer to the *Input Messages Manual*, 235-600-700, and the *Output Messages Manual*, 235-600-750, for additional information.

Admonishments None

Before you begin

Required Tools

- Use of Master Control Center (MCC) or Supplementary Trunk Line Workstation (STLWS).

Required Material

- The switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking feature software.

Required Information

- Refer to Office Record 5880 for provisioned GSMs.
- Refer to Office Record 5989 for provisioned IP routing data.

Required Conditions

- None required.

Procedure

1 Request a dump of IP routing tables for a specific SIP PH channel group number:

```
OP:TCPIP:RTDMP,CHNG=a-b-c-d;
```

Where:

a = Global switch module number

b = Packet switch unit 2 number

c = PSU2 shelf number

d = Channel group number

Result:

Generates a dump of the IP routing tables for a specific SIP PH channel group number.

If the result is `NG - NO ROUTE TABLE ENTRIES FOUND`, verify the IP routing table data on RC/V 33.3.

- If no data exists, then IP data has not been provisioned. Refer to the SIP for Packet Trunking Provisioning procedure, *IP Routing Tables*, 235-200-118.
- If data exists, this implies a trouble analysis state. SEEK TECHNICAL ASSISTANCE according to local procedures.

2 Stop. You have completed this procedure.

END OF STEPS

Dump Address Resolution Protocol (ARP) Table

Purpose

This procedure requests a dump of the ARP tables.

When to use

Use to troubleshoot layer 2 network connectivity problems between the *5ESS*® switch and the adjacent router.

Related information

Refer to the *Input Messages Manual*, 235-600-700, and the *Output Messages Manual*, 235-600-750, for additional information.

Admonishments

None

Before you begin

Required Tools

- Use of Master Control Center (MCC) or Supplementary Trunk Line Workstation (STLWS).

Required Material

- The switch must be running a 5E16.2 FR3 software release or later with the SIP for Packet Trunking feature software.

Required Information

- Refer to RC/V 5.80 for provisioned GSMs.

Required Conditions

- None required.

Procedure

1 Request a dump of the ARP table

`OP:TCPIP:ARPDMP,CHNG=a-b-c-d;`

Where:

a = Global switch module number

b = Packet switch unit 2 number

c = PSU2 shelf number

d = Channel group number

Result:

Generates a dump of the ARP tables for a specific channel group number.

If no entry is found, then do ping to the adjacent router:

```
EX:PING;
```

Reply should be returned. Dump ARP tables and if no entry is found, SEEK TECHNICAL ASSISTANCE according to local procedures.

2 Stop. You have completed this procedure.

END OF STEPS

□

Performance Monitoring on SIP PH

Overview

Performance monitoring is performed on the packet layers that terminate at the serving SIP PH. Information accumulated by the monitoring process enables the operational performance of the packet layers, used to transport SIP signaling calls, to be accurately assessed.

Performance monitoring is performed for the following packet layers:

- Ethernet
- IP
- Internet Control Message Protocol (ICMP)
- SCTP
- Session Initiation Protocol for Packet Trunking (SIPT)

Performance Measurements

Performance measurements are performed on packet layers that terminate on serving SIP PHs.

The basic interval for performance measurements is 15 minutes. A set of measurements is maintained for the following periods:

- current 15-minute period,
- previous 32 fifteen-minute periods, and
- current and previous 24-hour periods.

To detect excessive error rates automatically, thresholds are specified for the current 15-minute and 24-hour measurements.

Performance Parameters

Various parameters are used to quantify packet layers performance. The parameters are derived from the frequency of protocol errors detected at the serving SIP PH.

Reports

Violations of 15-minute and 24-hour thresholds are reported independently. Reports containing performance data are obtained using an MML command.

Alert Report Control

A packet layer threshold crossing alert (TCA) is reported whenever an error count crosses the specified threshold. TCA reports can be inhibited by using either RC/V or an MML command. Sets of parameters are defined to control the threshold crossing alerts that are reported independently by the performance monitoring process. Each set of parameters forms a threshold group.

The parameters specified in each threshold group are summarized as follows:

- flag to inhibit day threshold-crossing alert
- day threshold for each type of packet layer count
- flag to inhibit 15-minute threshold-crossing alert
- 15-minute threshold for each type of packet-error count

MML Commands

The following MML commands are used to control and retrieve SIP performance measurements:

- INIT:FAC
- OP:FAC
- ALW:FAC
- INH:FAC

Ethernet Layer Performance Monitoring Parameters

Overview

The following parameters are used to monitor Ethernet layer performance:

- Incoming Ethernet Frames Discarded
- Outgoing Ethernet Frames Discarded
- Invalid Ethernet Header
- MAC Sublayer Error
- Ethernet Unavailable Seconds

Incoming Ethernet Frames Discarded

This count keeps track of the total number of incoming Ethernet frames discarded due to the following error types:

- Unknown protocol
- Unsupported protocol
- Exceeds maximum permitted frame size
- Invalid frame check sequence

Outgoing Ethernet Frames Discarded

This count keeps track of the total number of outgoing Ethernet frames discarded.

Invalid Ethernet Header

This count keeps track of the total number of incoming invalid Ethernet header frames discarded.

- Unknown protocol
- Exceeds maximum frame size
- Invalid frame check sequence

MAC Sublayer Error

This count keeps track of the total number of incoming frames discarded due to MAC sublayer received and transmitted errors.

Ethernet Unavailable Seconds

This count is detected at the Ethernet layer. The Ethernet layer is considered to be unavailable whenever it is automatically inhibited for 10 consecutive seconds and continues to be considered unavailable until it has been automatically uninhibited for 10 consecutive seconds.

IP Layer Performance Monitoring Parameters

Overview

The following parameters are used to monitor IP layer performance:

- Invalid IP Header received
- Incoming IP Queue Congestion
- Invalid Fragmented IP Datagrams Received
- IP Unavailable Seconds

Invalid IP Header Received

This count keeps track of the total number of IP datagrams discarded due to the following error types:

- Invalid address in destination field
- Invalid version number
- Invalid header checksum
- Unknown or unspecified protocol

- Invalid header length
- Invalid total length

Incoming IP Queue Congestion

This count keeps track of the total number of IP datagrams discarded due to congestion in the incoming queue.

Invalid Fragmented IP Datagrams Received

This count keeps track of the total number of IP datagrams discarded because reassembly of fragmented packets could not be accomplished. Since reception of fragmented IP packets is a possible indication of a network or security issue, this count may be alarmed. The alarm level can be set to “no alarm” or “minor alarm”. The default is “no alarm”.

IP Unavailable Seconds

This count is detected at the IP layer. The IP layer is considered to be unavailable whenever the Ethernet layer is automatically inhibited for 10 consecutive seconds and continues to be considered unavailable until it has been automatically uninhibited for 10 consecutive seconds.

ICMP Layer Performance Monitoring Parameters

Overview

The following parameters are used to monitor ICMP layer performance:

- Invalid ICMP Datagrams Received
- ICMP Destination Unreachable Message Received
- ICMP Parameter Problem Message Received
- ICMP Echo Request Message Received
- ICMP TTL Exceeded Message Received
- ICMP Unavailable Seconds

Invalid ICMP Datagrams Received

This count keeps track of the total number of incoming ICMP datagrams discarded due to the following error types:

- Invalid/Unsupported ICMP message type
- Invalid ICMP checksum
- Invalid ICMP length
- Invalid ICMP code field

ICMP Destination Unreachable Message Received

This count keeps track of the total number of ICMP destination unreachable messages received.

ICMP Parameter Problem Message Received

This count keeps track of the total number of ICMP parameter problem messages received.

ICMP Echo Request Message Received

This count keeps track of the total number of ICMP echo request messages received. Since reception of numerous ICMP echo request messages can be a possible indication of a security issue, this count may be alarmed. The alarm level can be set to “no alarm” or “minor alarm”. The default is “no alarm”.

ICMP TTL Exceeded Message Received

This count keeps track of the total number of ICMP TTL exceeded messages received.

ICMP Unavailable Seconds

This count is detected at the ICMP layer. The ICMP layer is considered to be unavailable whenever the Ethernet layer is automatically inhibited for 10 consecutive seconds and continues to be considered unavailable until it has been automatically uninhibited for 10 consecutive seconds.

SCTP Layer Performance Monitoring Parameters

Overview

The following parameters are used to monitor SCTP layer performance:

- Invalid SCTP Header Received
- SCTP Chunk Errors Received
- SCTP Unavailable Seconds

Invalid SCTP Header Received

This count keeps track of the total number of incoming SCTP packets discarded due to the following SCTP common header error types:

- Unknown source port
- Unknown destination port

- Invalid verification tag
- Invalid checksum

SCTP Chunk Errors Received

This count keeps track of the total number of incoming SCTP packets discarded due to the following SCTP chunk error types:

- Invalid stream identifier
- Missing mandatory parameter
- Stale cookie error
- Out of resource
- Unresolvable address
- Unrecognized chunk type
- Invalid mandatory parameter
- Unrecognized parameters
- No user data
- Cookie received while shutting down

SCTP Unavailable Seconds

This count is detected at the SCTP layer. The SCTP layer is considered to be unavailable whenever the Ethernet layer is automatically inhibited for 10 consecutive seconds and continues to be considered unavailable until it has been automatically uninhibited for 10 consecutive seconds.

SIPT Layer Performance Monitoring Parameters

Overview

The following parameters are used to monitor SIPT layer performance:

- SIPT Messages Discarded
- SIPT Unavailable Seconds

SIPT Messages Discarded

This count keeps track of the total number of SIPT messages discarded due to the following error types:

- Unknown/unsupported Protocol version
- Unknown/unsupported SIP method
- Unknown/unsupported Request-URI

SIPT Unavailable Seconds

This count is detected at the SIPT layer. The SIPT layer is considered to be unavailable whenever the Ethernet layer is automatically inhibited for 10 consecutive seconds and continues to be considered unavailable until it has been automatically uninhibited for 10 consecutive seconds.

New Trouble Isolation Techniques

Refer to Corrective Maintenance Procedures.

□

Glossary

A

ACM
Address Complete

ADM
Add/Drop Multiplexer

AM
Administrative Module

ANSI
American National Standards Institute

AP
Administration Processor

APS
Automatic Protection Switching

ARP
Address Resolution Protocol

ASM
Administrative Services Module

AT
Access Tandem

B

BER
Basic Encoding Rules/Bit Error Rate

BHC
Busy Hour Call

BNID
Bearer Network ID

C **CCITT**
Consultative Committee on ITT

CFA
Carrier Facility Alarm

CIC
Circuit/Call Identification Code

CIDR
Classless Interdomain Routing

CLI
Change Level Indicator/Circuit Level Indicator/Call Line Identification/Common Level Indicator

CM
Communications Module

CMP
Communications Module Processor

CPE
Customer Premises Equipment

CPG
Call Progress

CRC
Cyclic Redundancy Check

CSC
Common Signalling Channel

CSMA/CD
Carrier Sense Multiple Access with Collision Detection

D **DF**
Do Not Fragment

DF2
Data Fanout 2

DIFFSERV
Differentiated Service

DNS
Domain Name Server

DNUS
Digital Network Unit - SONET

DPM
Digital Performance Monitoring

DRM
Distinctive Remote Module

DS-0/1
Digital Signal-Level 0 (64 kb/s)/DS-1 Digital Signal-Level 1 (1.544 Mb/s)

DSCP
Differentiated Services Code Point

DSP
Digital Signal Processor

DTMF
Dual Tone Multifrequency

E

EC
Echo Cancellation (or canceller)

ECN
Explicit Congestion Notification

EMS
Element Management System

EO
End Office

EXM
Exit Message

F **FAC**
Facility

FE
Frame Error or Far End

FIT
Failure in Time

FPGA
Field Programmable Gate Array

FSD
Feature Specifications Document

G **G711**
PCM of Voice Frequency Standard

GA
General Availability

GQPH
General QLPS Protocol Handler

GR
Generic Requirements

GSM
Global Switching Module

GTWY
Gateway

H **HDLC**
High Level Data Link Control

HW
Hardware

I

I/O
Input/Output

IAM
Initial Address Message

ICMP
Internet Control Message Protocol

ILEC
Incumbent Local Exchange Carrier

INH
Inhibit

IOSE
IP OIU-Packet SIP EQPOTS

IOST
IP OIU-Packet SIP Tandem

IP
Internet Protocol

IPM
IP Protocol Monitoring

IPv4/6
Internet Protocol Version 4/Version 6

ISUP
ISDN User Part

ITU
International Telecommunications Union

ITU-T
ITU-Telecommunications Standardization Sector (formally CCITT)

IXC
Interexchange Carrier

L **LAN**
Local Area Network

LEC
Local Exchange Carrier

LSB
Least Significant Bit

LT
Local Tandem

LTE
Line Terminating Equipment

M **MCC**
Master Control Center

MF
Multifrequency or More Fragments

MH
Message Handler

MHz
Mega Hertz (1 million cycles per second)

MIME
Multipurpose Internet Mail Extension

MSB
Most Significant Bit

N **NAR**
North American Region

NCT2
Network Control Timing Link 2

NIC
Network Interface Card

NM
Network Management

NP
Network Protocol

O

OAM&P
Operations, Administration, Maintenance and Provisioning

OC
Optical Carrier

OC-3c
Optical Carrier - level 3 concatenated

OFI
Optical Facility Interface

OFI-IP
Optical Facility Interface-Internet Protocol

OIU
Optical Interface Unit

OIU-IP
Optical Interface Unit-Internet Protocol

OOS
Out of Service

OPS
Originating Packet Switch

OSI
Open System Interface

OSS
Operation Support System

P

PCT
Peripheral Control and Timing Link

PH
Protocol Handler

PHB
Per Hop Behavior

PLI
Peripheral Link Interface

PM
Performance Monitoring

POS
Packet Over SONET

PPP
Point-to-Point Protocol

PPPLK
PPP Link

PS
Packet switching Subsystem

PSTN
Public Switched Telephone Network

PSU2
Packet Switch Unit 2

Q

QLPS
Quad Link Packet Switch

QOS
Quality Of Service

QPH
QLPS Protocol Handler

R

RC/V
Recent Change and Verify

RFC
Request for Comments

RS
Redirect Server

RSW
Resident Software

RTA
Routing and Terminal Administration Subsystem

RTO
Retransmission Timeout

RTP
Real Time Transfer Protocol

RTT
Round Trip Time

S

SAC
Synchronous to Asynchronous Conversion

SACK
Selective Acknowledgement

SCTP
Stream Control Transmission Protocol

SDL
Signalling Data Link

SDP
Session Description Protocol

SFID
Secured Feature ID

SIP
Session Initiation Protocol

SIP PH
SIP Protocol Handler

SIP-I
Session Initiation Protocol with Encapsulated ISUP

SIP-T
Session Initiation Protocol - Telephony

SM
Switching Module

SMP
Switching Module Processor

SONET
Synchronous Optical Network

SPE
SONET Payload Envelope/Synchronous Payload Envelope

SS7
Signalling System 7

SSRC
Synchronization Source

SSTR
Service Selection Trunk Reservation

STE
SONET Terminating Equipment (also, RSTE: Regenerator Section Terminating Equipment)

STP
Signal Transfer Point

SU
Software Update

SW
Software

T

TAS
Telephone Application Server

TCAP
Transaction Capabilities Application Part

TCB
Transmission Control Block

TCP
Transmission Control Protocol

TDM
Time Division Multiplexing

TE
Termination Equipment

TGC
Trunk Group Control

TOS
Type of Service

TPS
Terminating Packet Switch

TR
Trunk Reservation

TSI
Time Slot Interchange

TSN
Transmission Sequence Number

TTL
Time to Live

U

UAC
User Agent Client

UAS
User Agent Server

UDP
User Datagram Protocol

URI
Universal Resource Identifier

V **VCDX**
Very Compact Digital Exchange

VLSM
Variable Length Subnet Masking

VoIP
Voice Over Internet Protocol

VoP
Voice Over Packet

Index