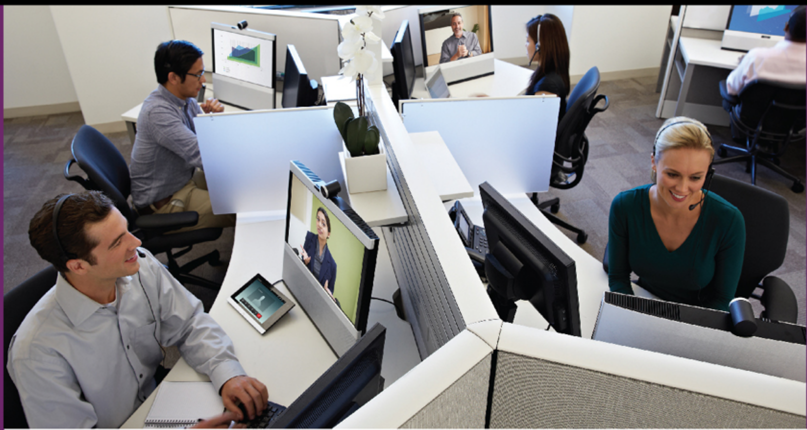
IP COMMUNICATIONS



Understanding Session Border Controllers

Comprehensive Guide to Designing, Deploying, Troubleshooting, and Maintaining Cisco Unified Border Element (CUBE) Solutions

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Foreword by **Jonathan Rosenberg**

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RFC 3665, "Session Initiation Protocol (SIP) Basic Call Flow Examples," https://tools.ietf.org/html/rfc3665

RFC 3842, "A Message Summary and Message Waiting Indication Event Package for the Session Initiation Protocol (SIP)," https://tools.ietf.org/html/rfc3842

RFC 3891, "The Session Initiation Protocol (SIP) 'Replaces' Header," https://tools.ietf.org/html/rfc3891

RFC 3892, "The Session Initiation Protocol (SIP) Referred-By Mechanism," https://tools.ietf.org/html/rfc3892

RFC 3911, "The Session Initiation Protocol (SIP) 'Join' Header," https://tools.ietf.org/html/rfc3911

RFC 3960, "Early Media and Ringing Tone Generation in the Session Initiation Protocol (SIP)," https://tools.ietf.org/html/rfc3960

RFC 4028, "Session Timers in the Session Initiation Protocol (SIP)," https://tools.ietf.org/html/rfc4028

RFC 4488, "Suppression of Session Initiation Protocol (SIP) REFER Method Implicit Subscription," https://tools.ietf.org/html/rfc4488

RFC 4826, "Extensible Markup Language (XML) Formats for Representing Resource Lists," https://tools.ietf.org/html/rfc4826

RFC 5118, "Session Initiation Protocol (SIP) Torture Test Messages for Internet Protocol Version 6 (IPv6)," https://tools.ietf.org/html/rfc5118

RFC 5363, "Framework and Security Considerations for Session Initiation Protocol (SIP) URI-List Services," https://tools.ietf.org/html/rfc5363

RFC 5364, "Extensible Markup Language (XML) Format Extension for Representing Copy Control Attributes in Resource Lists," https://tools.ietf.org/html/rfc5364

RFC 5368, "Referring to Multiple Resources in the Session Initiation Protocol (SIP)," https://tools.ietf.org/html/rfc5368

RFC 5589, "Session Initiation Protocol (SIP) Call Control—Transfer," https://tools.ietf.org/html/rfc5589

RFC 5768, "Indicating Support for Interactive Connectivity Establishment (ICE) in the Session Initiation Protocol (SIP)," https://tools.ietf.org/html/rfc5768

RFC 5853, "Requirements from Session Initiation Protocol (SIP) Session Border Control (SBC) Deployments," https://tools.ietf.org/html/rfc5853

RFC 5923, "Connection Reuse in the Session Initiation Protocol (SIP)," https://tools.ietf.org/html/rfc5923

RFC 6157, "IPv6 Transition in the Session Initiation Protocol (SIP)," https://tools.ietf.org/html/rfc6157

RFC 6228, "Session Initiation Protocol (SIP) Response Code for Indication of Terminated Dialog," https://tools.ietf.org/html/rfc6228

RFC 6337, "Session Initiation Protocol (SIP) Usage of the Offer/Answer Model," https://tools.ietf.org/html/rfc6337

RFC 6665, "SIP-Specific Event Notification," https://tools.ietf.org/html/rfc6665

RFC 7118, "The WebSocket Protocol as a Transport for the Session Initiation Protocol (SIP)," https://tools.ietf.org/html/rfc7118

RFC 7584, "Session Traversal Utilities for NAT (STUN) Message Handling for SIP Back-to-Back User Agents (B2BUAs)," https://tools.ietf.org/html/rfc7584

RFC 7614, "Explicit Subscriptions for the REFER Method," https://tools.ietf.org/html/rfc7614

RFC 7647, "Clarifications for the Use of REFER with RFC 6665," https://tools.ietf.org/html/rfc7647

RFC 7984, "Locating Session Initiation Protocol (SIP) Servers in a Dual-Stack IP Network," https://tools.ietf.org/html/rfc7984

draft-klatsky-dispatch-ipv6-impact-ipv4-03, "Interoperability Impacts of IPv6 Interworking with Existing IPv4 SIP Implementations," https://tools.ietf.org/html/draft-klatsky-dispatch-ipv6-impact-ipv4-03

H.225, "Call Signalling Protocols and Media Stream Packetization for Packet-Based Multimedia Communication Systems," https://www.itu.int/rec/T-REC-H.225.0-200912-I/en

H.245, "Control Protocol for Multimedia Communication," https://www.itu.int/rec/T-REC-H.245/en

E.180, "Technical Characteristics of Tones for the Telephone Service," https://www.itu.int/rec/T-REC-E.180-199803-I/en

"Various Tones Used in National Networks (According to ITU-T Recommendation E.180)," https://www.itu.int/ITU-T/inr/forms/files/tones-0203.pdf

ETS 300 512, "European Digital Cellular Telecommunications System (Phase 2); Procedure for Call Progress Indications (GSM 02.40)," http://www.etsi.org/deliver/etsi_i_ets/300500_300599/300512/01_60/ets_300512e01p.pdf

"Delayed Offer to Early Offer Translation for SIP Calls," https://www.cisco.com/c/en/us/td/docs/ios/voice/cube/configuration/guide/vb_8241.html

"Delayed-Offer to Early-Offer," https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube/configuration/cube-book/voi-cube-do-eo.pdf

"SIP Profiles," https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube_fund/configuration/xe-3s/cube-fund-xe-3s-book/voi-sip-param-mod.html

"Multicast Music-on-Hold Support on Cisco UBE," https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube_proto/configuration/15-mt/cube-proto-15-mt-book/voi-multicast-moh.html

"Dynamic Refer Handling," https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube_sip/configuration/15-mt/cube-sip-15-mt-book/voi-cub-sip-dyn-refer-handling.html

"Configuring SIP Call-Transfer Features," https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/sip/configuration/15-mt/sip-config-15-mt-book/voi-sip-call-transfer.html

"Mid-call Signaling Consumption," https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube/configuration/cube-book/voi-cube-midcall-reinvite.html

"Nano CUBE SUBSCRIBE-NOTIFY Passthrough," https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube_nano/configuration/15-mt/nanocube-config-15-mt-book/voi-nanocube-subscribe-notify.html

H.323-to-SIP Interworking on CUBE," https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube/configuration/cube-book/h323-to-sip.html

"H.323 to SIP Interworking," https://www.cisco.com/c/en/us/td/docs/routers/asr1000/configuration/guide/sbcu/2_xe/sbcu_2_xe_book/sbc_hsp.pdf

"Configurable Pass-Through of SIP INVITE Parameters," https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube_sip/configuration/15-mt/cube-sip-15-mt-book/voi-conf-pass-thro.html

"Configuring Support for Dynamic REFER Handling on the Cisco UBE," https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube_sip/configuration/xe-3s/cube-sip-xe-3s-book/voi-cub-sip-dyn-refer-handling.pdf

"Achieving SIP RFC Compliance," https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/sip/configuration/15-mt/sip-config-15-mt-book/voi-sip-rfc.html

"SIP Profiles," https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube/configuration/cube-book/voi-sip-param-mod.html

"VoIP for IPv6," https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube/configuration/cube-book/voi-cube-voip-ipv6.pdf

"ICE-Lite Support on CUBE," https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube/configuration/cube-book/voi-cube-ice-lite.html

"Nano CUBE SUBSCRIBE-NOTIFY Passthrough," https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube_nano/configuration/15-mt/nanocube-config-15-mt-book/voi-nanocube-subscribe-notify.html

Chapter 5

Media Processing

IP networks form the foundation for exchanging voice and video packets. There has to be a set of rules and procedures to provide an overarching framework for transmission of real-time media information between participants in a communication session. Real-Time Transport Protocol (RTP) was developed for precisely this reason, and it has seen overwhelming acceptance across the industry. RTP and many of its extensions have been built and standardized over many years to fit a myriad of different situations.

SBCs are devices that have operations defined in both the media and signaling planes, with the ability to transform signaling and media and to strongly influence session characteristics. Over the years, SBCs have been developed and installed in real-time multimedia networks to "fix" some of the inherent interoperability challenges seen when attempting to establish multimedia sessions across network boundaries.

This chapter talks about the various media plane operations that are performed by SBCs and also discusses certain considerations that need to be taken into account with media handling by SBCs.

This chapter includes the following sections:

- **Real-Time Transport Protocol—**This section provides an introduction to Real-Time Transport Protocol (RTP) and explains how RTP packets are formatted.
- **Real-Time Transport Control Protocol—**This section covers Real-Time Transport Control Protocol (RTCP) as a peer protocol to RTP and discusses various RTCP packet types.
- **SBC Handling of RTP and RTCP—**This section discusses how SBCs handle RTP and RTCP traffic, as well as the different considerations that arise when SBCs manipulate RTP and RTCP packets.

- **Symmetric and Asymmetric RTP/RTCP—**This section covers the symmetric and asymmetric properties of RTP.
- **DSP-Based RTP Handling on SBCs—**This section covers the various RTP/RTCP manipulation operations that can be carried out by digital signal processors (DSPs).
- **Media Anti-tromboning—**This section covers how media loops are created on SBCs and discusses techniques for mitigating such media loops.
- **Alternate Network Address Types—**This section covers how SBCs can work with multiple address types (IPv4 and IPv6) when setting up a communication session.
- **Solving NAT Traversal Challenges—**This chapter covers the basics of Network Address Translation (NAT), the problems NAT introduces in real-time media transmission, and some of techniques that can help overcome these problems.
- **Troubleshooting RTP—**This section includes diagnostic commands, debugging snippets, and a general methodology for troubleshooting media-related issues on SBCs, using CUBE as an example.

By the end of this chapter, you will have a thorough understanding of RTP and RTCP, how SBCs handle and modify RTP and RTCP traffic, and a general approach for troubleshooting RTP issues on SBCs.

Real-Time Transport Protocol

Real-Time Transport Protocol (RTP), originally defined in RFC 1889 and superseded by RFC 3550, provides a framework for the end-to-end transport of voice and video. RTP typically operates over UDP/IP and provides built-in loss detection, receiver feedback, source identification, important event indications, and sequencing. RTP has a peer protocol, Real-Time Control Protocol (RTCP), that provides media reception feedback for the related RTP stream. RTCP is discussed in further detail in the following section.

Central to the operation of RTP is the concept of an RTP session. An RTP *session* is a group of participants interacting over RTP, such that a given participant may be a part of several different RTP sessions at the same time. For example, a pair of endpoints could have both an audio RTP session and a video RTP session active between them. An RTP session is identified by the combination of a network address and port pair on which traffic is sent and received. Different ports may be used for RTP and RTCP for each session.

An RTP session can be either unicast (one-to-one communication between a pair of participants) or multicast (one-to-many communication to participants). Before exploring various other topics discussed in this and subsequent chapters, it is important to first take a close look at the RTP packet format.

RTP Packet Format

An RTP packet consists of two parts: an RTP header and an RTP payload (with optional padding). Figure 5-1 shows the RTP packet format.

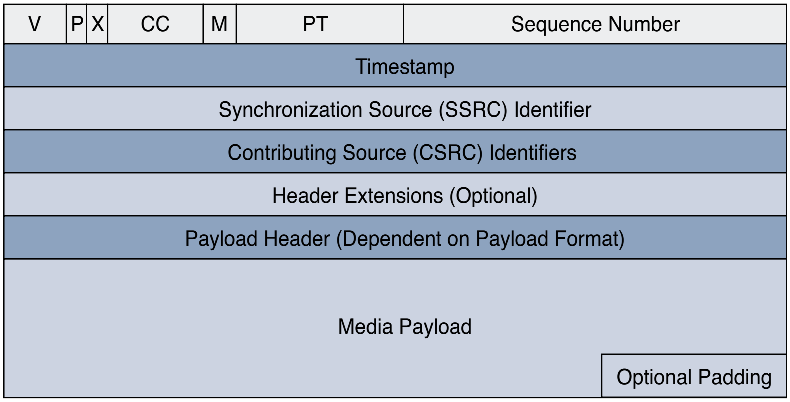

Figure 5-1 *RTP Packet Format*

Descriptions of the various fields appearing in the RTP header are listed:

- **Version (V)**—This header field specifies the RTP version in use. The current version at the time of this publication is 2.
- **Padding (P)**—This single-bit header field, when set to 1, indicates that there are additional octets appended to the RTP payload. These additional octets are not a part of the payload and are primarily inserted to ensure that certain encryption algorithms always work on fixed-size blocks of data.
- **Extension (X)**—This single-bit field indicates the presence of an RTP header extension. RTP header extensions are required to carry additional media session information that cannot be encoded within the standard RTP headers or payload. Typical examples of this include the RTP header extensions for audio level information of RTP samples, as defined in RFC 6464. Although header extensions are not commonly implemented, it is important for the specification to have such accommodation for these rare cases.
- **CSRC count (CC)**—This field identifies the number of CSRC identifiers that follow the fixed header field. CSRCs are explained further later in this section.
- **Marker bit (M)**—This header filed is used to designate important events during the media session. For example, the marker bit might designate the start of a new DTMF event. This usage can be observed when using named telephony events for DTMF transmission. (See Chapter 7, "DTMF Interworking," for more details.) Yet another usage of the M field is when the payload format changes during a media session. For example, a media session might negotiate G.711 as the audio codec and begin transmission of RTP packets back and forth. Sometime during the course of the communication session, an application interaction might cause the audio codec to change