SAP BCM and VoIP Technology – A Technical Overview

Applies to:

SAP Business Communications Management 6.0 and higher. For more information visit SAP Business Communications Management.

Summary

The SAP BCM implementation and voice channel deployment is a big challenge due to the different concepts and protocols involved in VoIP technology. This article describes the main protocols used to establish a VoIP call applied to SAP BCM including practical examples and a technical overview of VoIP packets over the network.

Author: Heber Olivar Silva

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Author Bio

Heber Olivar Silva is a SAP BCM Consultant and VoIP technology specialist (Cisco CCVP certified) with 9 years experience with legacy telephony systems, Voice over IP, Unified Communications and contact center solutions.

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SAP BCM Overview

SAP Business Communications Management (SAP BCM), one of the newest members of SAP CRM solution, allows companies to improve their Contact Center areas and deploy communication-enabled business processes. The SAP BCM solution provides multi-channel integration, such as e-mail (push mode), instant messaging, SMS and telephony to be integrated with SAP CRM.

The telephony channel provided by the BCM uses VoIP technology, also known as IP Telephony. Beside the large portfolio of SAP solutions and technologies, the use of VoIP technology require a extra knowledge about networks and telecommunication due to the numerous technical details and protocols involved.

VoIP technology allows voice communication using the Ethernet network and on the Internet, allowing CSR mobility and a contact center solution with multiple sites, contingency and redundancy scenarios contributing to the cost optimization.

System Landscape

Starting at the SAP BCM landscape is necessary to understand and plan how to deliver a phone call, often still using the TDM (time-division multiplexing) traditional telephony. First of all you need to understand that SAP BCM is not a CTI connector available from SAP to connect with other CTI solutions, the BCM is a comprehensive IP communications system such as IP-PBX that allows the use of VoIP not only in the contact centers but also to all users who wish to enjoy the benefits of mobility that technology permits.

Actually some Telecom operators already offer to their customers a direct VoIP connection, but mostly the interface connection between provider and customer is still done through TDM traditional telephony (E1 - Europe standard and T1 - USA standard) with signaling protocols such as CAS and CCS. These communication interfaces are not connected directly to the BCM being necessary convert these traditional interfaces and protocols for VoIP standards. This conversion is done by so-called VoIP gateways or PBX that supports VoIP technology and can act as a gateway and still allowing access to back-office departments using PBX infrastructure.

BCM adopts the two main VoIP signaling protocols currently used by telecom manufactures allowing interconnectivity with third party equipment (gateway or pbx voip-enabled). The connection can be made using the signaling protocols H.323 and SIP and voice codecs G.729 and G.711.

Figure 1: Using a VoIP Gateway

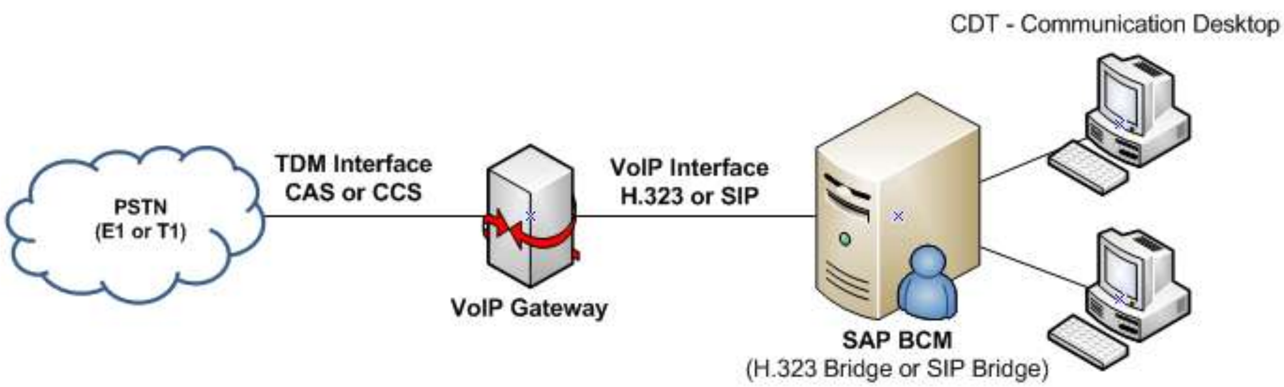
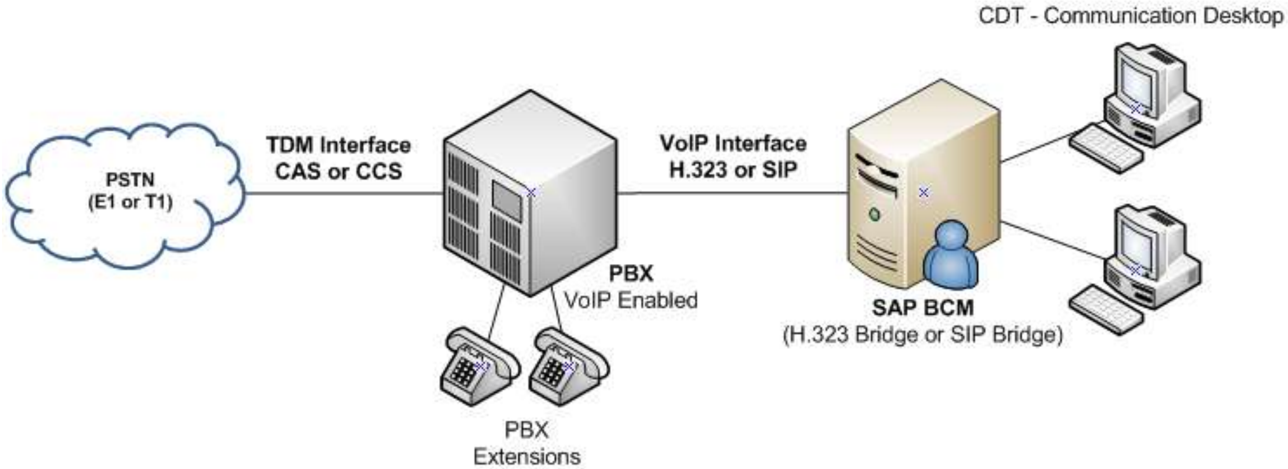


Figure 2: Using a PBX VoIP-enabled



VoIP Signaling protocols

The main goal of VoIP signaling protocols is create, manage and terminate a bidirectional Real-time Transport Protocol (RTP) stream between endpoints involved in a conversation.

H.323 protocol stack

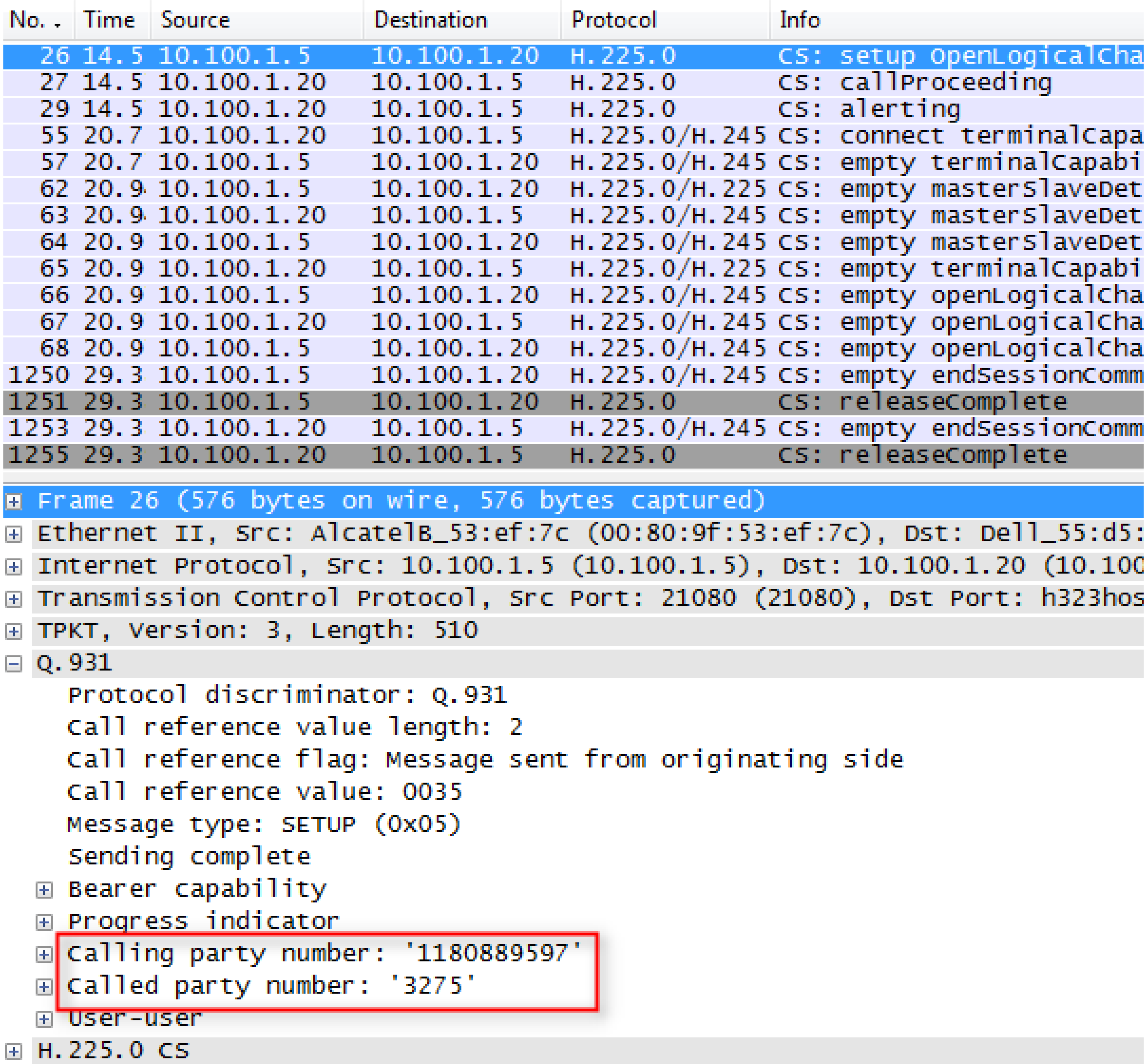
The H.323 is a suite of protocols defined by the International Telecommunication Union (ITU) and actually is the most widely deployed voice protocol. The protocols specified by H.323 include:

H.225.0 Call Signalling (Q.931, ISDN signaling) is used to establish a connection between two H.323 systems and endpoints.

H.225 Registration, Admission, and Status (RAS) is used between endpoints (terminal and gateways) to perform registration, admission control, bandwidth changes, status, and disengage procedures between endpoints.

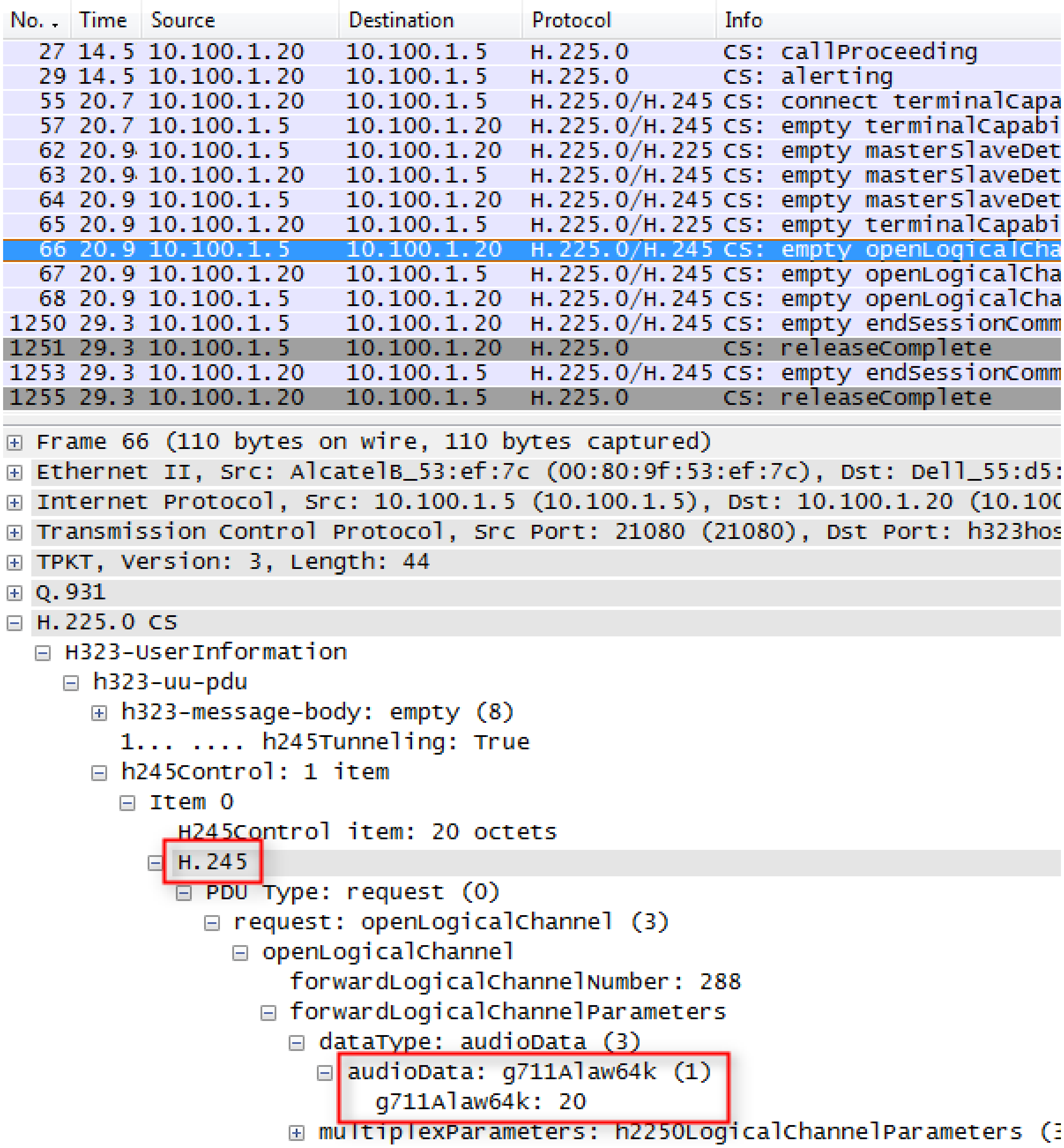
H.245 Control Signaling is used to exchange end-to-end control messages regarding the operation such as capabilities exchange, opening and closing of logical channels used to carry media streams and flow-control messages.

Figure 3: H.323 call flows (H.225.0)



Note: Analyzing package generated by H.225.0 protocol is possible to identify the parameters “Calling Party Number” and “Called Party Number” with the parties involved in this connection request.

Figure 4: H.323 call flows (H.245)

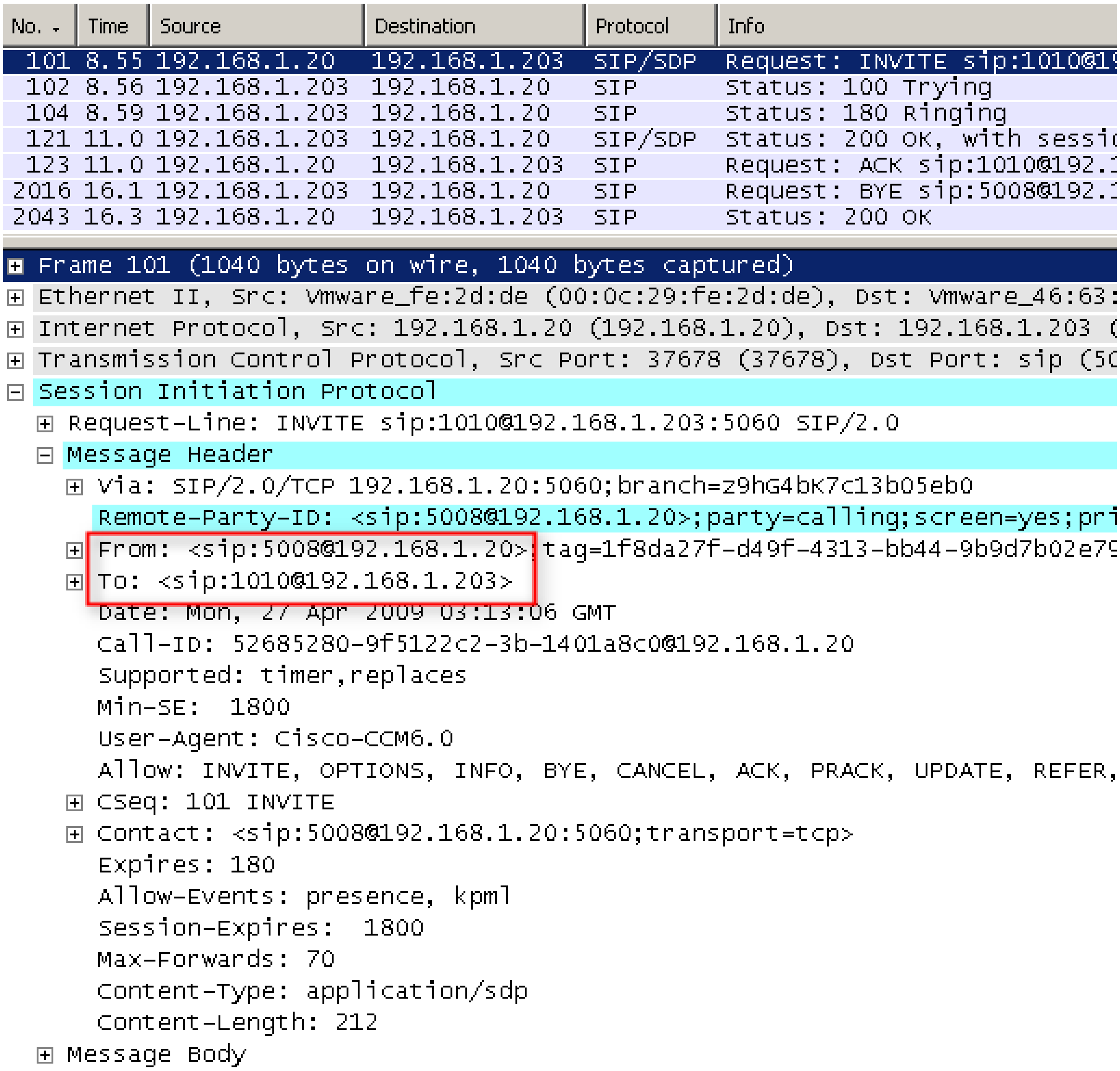


Note: Analyzing package generated by H.245 protocol is possible to identify the parameters “audioData” with the codec negotiated in this connection and “g711Alaw64k” with the value indicating the voice sample size.

SIP protocol – Session Initiation Protocol

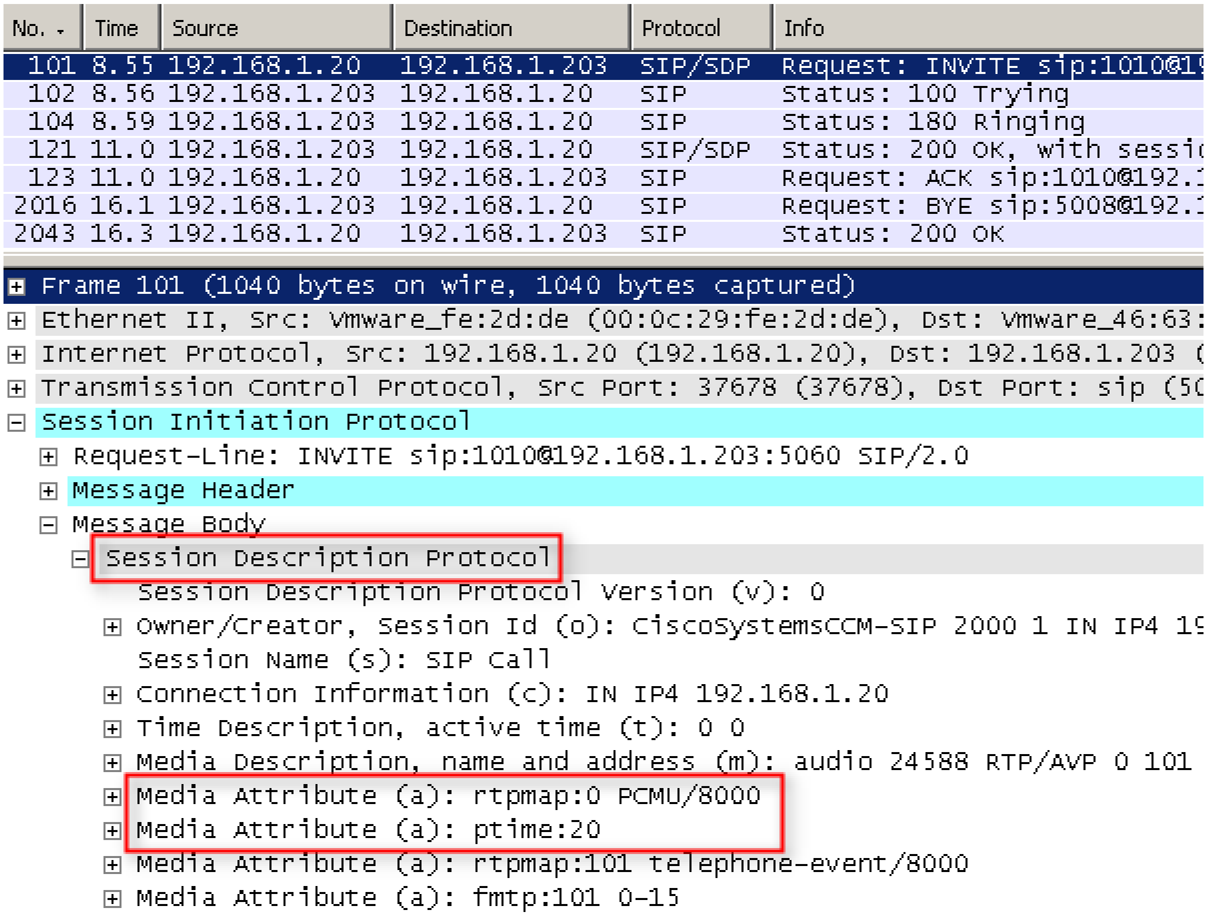
SIP is a protocol developed by Internet Engineering Task Force (IETF) and compliant with the following standards RFC 2543, RFC 3261 and RFC 3665. SIP uses ASCII-text-based messages to communicate and you can implement and troubleshoot very easy if compared with H.323. SIP is a protocol that can be used with other IETF protocols to build a complete multimedia architecture such as Session Description Protocol (SDP) for describing multimedia sessions.

Figure 5: SIP call flows (INVITE msg)



Note: Analyzing package generated by SIP protocol is possible to identify the parameters “From” and “To” with the parties involved in the INVITE message.

Figure 6: SIP call flows (SDP parameters)

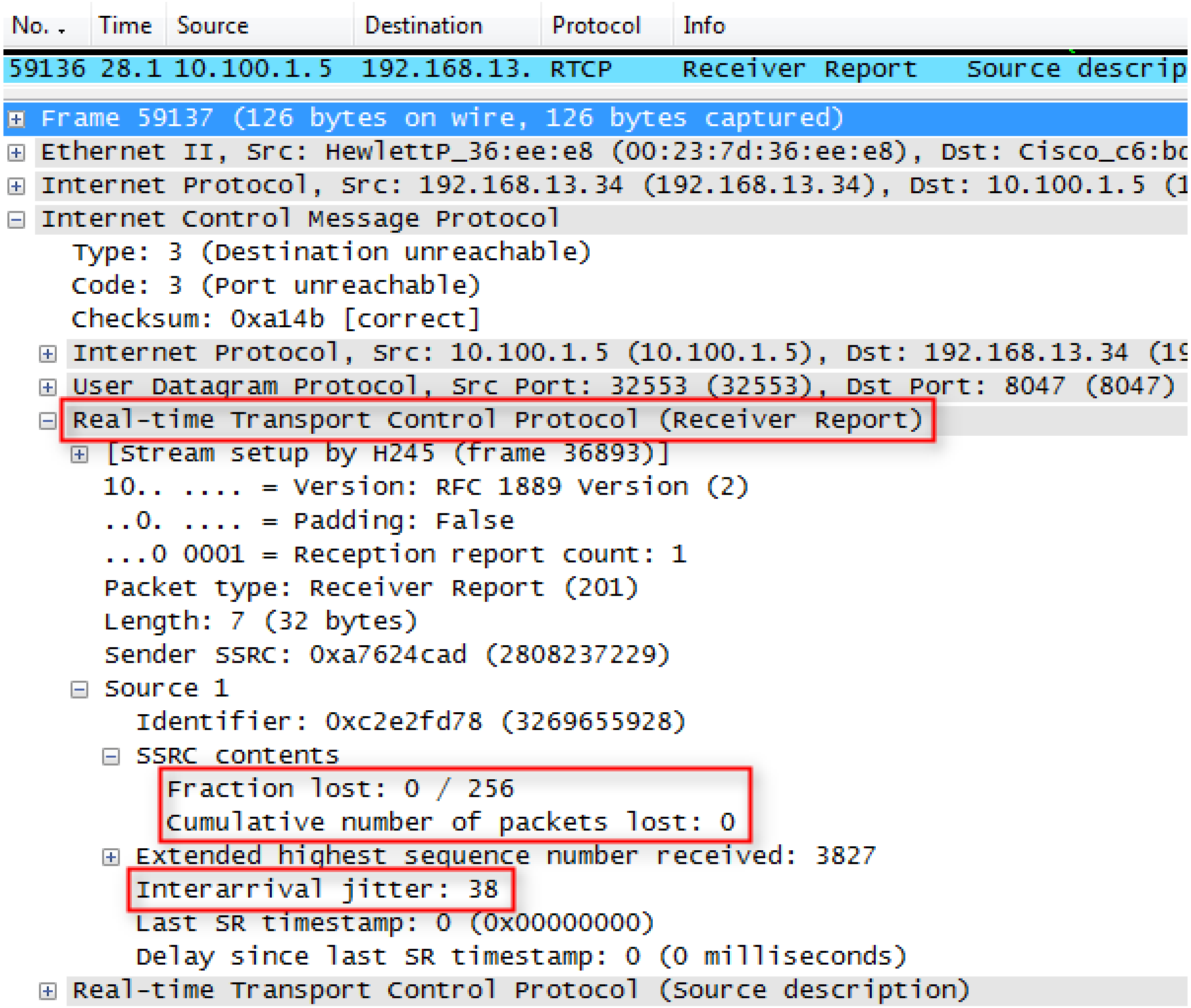


Note: Analyzing package generated by SIP/SDP protocol is possible to identify the parameters "Media Attribute (a): rtpmap" with the codec negotiated in this connection (0 PCMU/8000 is G.711 according to standard) and "Media Attribute (a): ptime" with the value indicating the voice sample size.

RTP and RTCP protocols

In a VoIP network, the voice data are transported using RTP according to standards RFC 1889 and RFC 3550 that define packet format for delivering audio and video over the internet. The RTCP is an auxiliary protocol to RTP that provides information for RTP streams but does not transport any voice data and used for QoS reporting gathering statistics such as bytes sent, packets sent, lost packets, jitter, feedback and round-trip delay.

Figure 7: RTCP protocol



Note: Analyzing package generated by RTCP protocol is possible to identify the packets lost and inter-arrival jitter.

Codecs

A codec performs encoding and decoding of a digital stream. It is important to consider which codec will be deployed and prepare the correct capacity planning to network bandwidth. Coding techniques are standardized by ITU and there are several types, but we will focus in G.729 and G.711 that are supported by SAP BCM.

G.711 encoding telephone audio on a 64 kbps channel without compression and offers toll-quality voice conversations at the cost of bandwidth consumption and is suited mainly to be deployed in LAN environments.

G.729 encoding telephone audio on an 8 kbps channel with compression and offers a reduction in bandwidth consumption at the cost of near toll-quality voice conversations and is suited mainly to be deployed in WAN environments.

Voice sample size is a variable that can affect total bandwidth used and must be considered in a design phase because of the voice sample size used to build voice packet influences direct on packet sizes and the necessary bandwidth. Setting more voice samples in a voice packet, the packets are larger and the bandwidth is reduced, but the risk to transport this packet over the network is bigger and excessive delays and packet loss may happen. The BCM default value is 30 ms, but you can change if it is necessary.

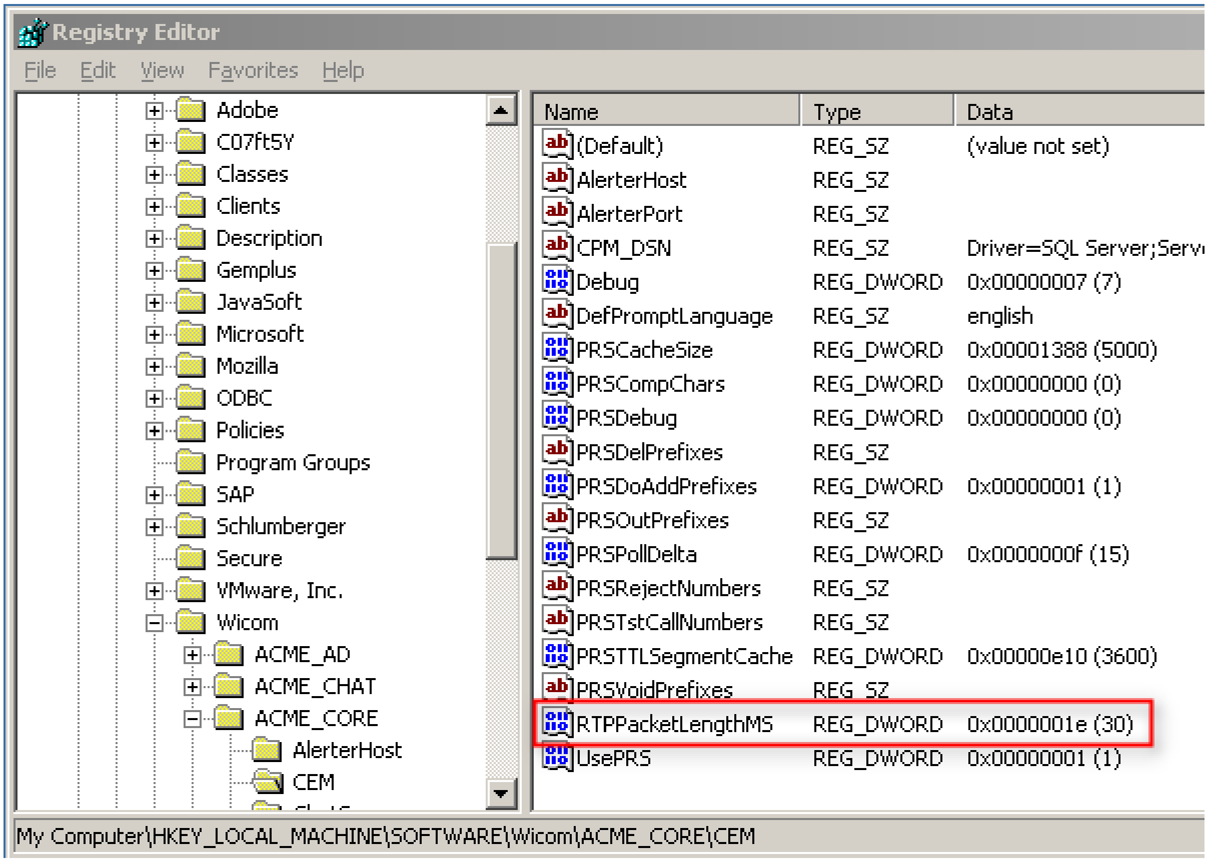
Table 1: Packets to transmit on second of conversation.

Codec	Bandwidth (kbps)	Sample Size (ms)	Sample Size (Bytes)	Packets Per Second
G.711	64	30	240	33
G.711	64	20	160	40
G.729	8	30	40	25
G.729	8	20	20	50

To determine the number of bytes encapsulated in a packet based on the codec bandwidth and sample size use the following formula:

$$\text{Bytes per Sample (Bytes)} = \frac{\text{Sample Size (ms)} * \text{Codec Bandwidth (Kbps)}}{8}$$

Figure 8: Adjusting BCM packet length



Note: It's possible adjust the packet length using *Registry Editor* in CEM Server and changing the value to DWORD RTPPacketLengthMS. BCM default value is 30 ms.

VoIP bandwidth per call

The total bandwidth necessary to ensure VoIP traffic takes into account data-link header, IP header, UDP header, RTP header, voice codec and sample size.

Consider the following values:

- Ethernet header = 18 bytes
- IP header = 20 bytes
- UDP header = 8 bytes
- RTP header = 12 bytes
- Voice payload to G.711 = 160 bytes, G.729 = 20 bytes

To determine the total bandwidth per VoIP calls use the following formula:

$$\text{Total Bandwidth (Kbps)} = \frac{\text{Layer 2 Overhead (Bytes)} + \text{IP UDP RTP Overhead (Bytes)} + \text{Sample Size (ms)}}{\text{Sample Size (ms)}} * \text{Codec Speed (Kbps)}$$

$$\text{Total Bandwidth (Kbps)} = \frac{18 + 40 + 20}{20} * 8$$

Total Bandwidth (Kbps) = 31.2 Kbps (per call using G.729 codec)

Note: Note that protocols headers influence the data bandwidth required. For G.729 VoIP call is not only 8 Kbps but 31.2Kbps.

Quality of Services – QoS

Configuring voice in a data network requires network services with low delay, minimal jitter, and minimal packet loss. The necessary bandwidth must be calculated based on the codec used and the number of concurrent connections. QoS must be configured to minimize jitter and loss of voice packets.

Jitter: Jitter is a variation in the arrival of coded speech packets in a VoIP network.

Delay: Delay is the time spent between the spoken voice and the arrival of the voice packet at the endpoint that results from multiples factors such as distance, coding, compression, serialization and buffers. According with ITU-T G.114 recommendation the value acceptable for most user applications is between 0 and 150 ms.

Packet loss: Lost packets are not recoverable (RTP/UDP protocol characteristic) resulting in gaps in the conversation caused by unstable network, network congestion, and too much variable delay.

QoS tools

Real-time applications have different characteristics and requirements from traditional data applications, therefore voice applications tolerate minimal variation in delay, packet loss and jitter. To effectively transport VoIP traffic, mechanisms are required to ensure reliable delivery of voice packets know as QoS techniques. In summary QoS features implement the following services:

- Guaranteed bandwidth: Ensure that necessary bandwidth is always available to support voice and data traffic.
- Avoid network congestion: Ensure that LAN and WAN infrastructure can support the traffic volume.
- Shape network traffic: Traffic-shapping tools ensures smooth and consistent delivery of frames over the network.
- Set traffic priorities across the network: Mark voice packets as priority and routes to the right priority queue.

Differentiated Services – DSCP

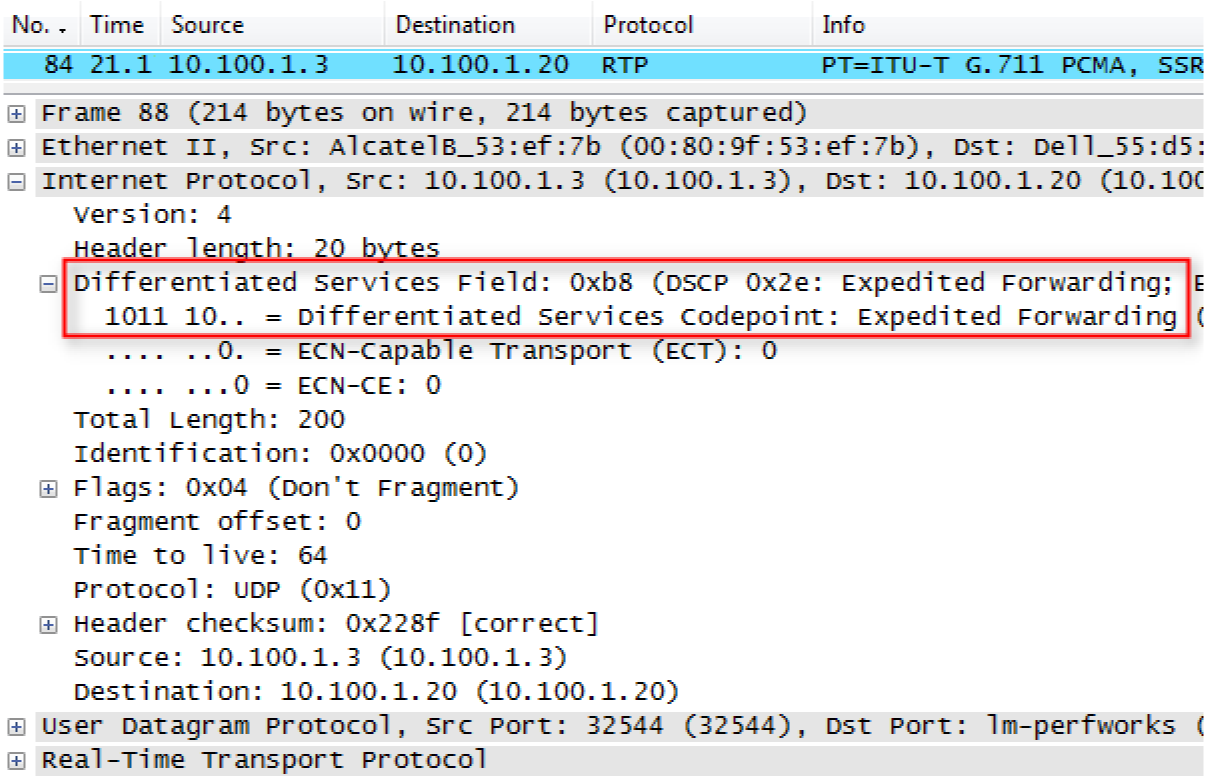
Differentiated Services, known as DiffServ, consist in a mark in the packets at moment that they ingress into a network and permit that network devices QoS-enabled can evaluate this mark relate with the class of service and do the right choice to route them. To permit this marking in a multimedia network, the IP header has been redefined to include a 6-bit Differentiated Services Code Point (DSCP) field (RFC 2474, RFC 2475, RFC 2597).

Expedited Forwarding and DSCP Values

The RFC 2598 defines the expedited forwarding behaviors that simply states that a packet with the EF DSCP should minimize delay, jitter and loss, up to a guaranteed bandwidth level and suggests that a QoS action must be performed like queuing tools to minimize the time that EF packets spend in a "priority queue".

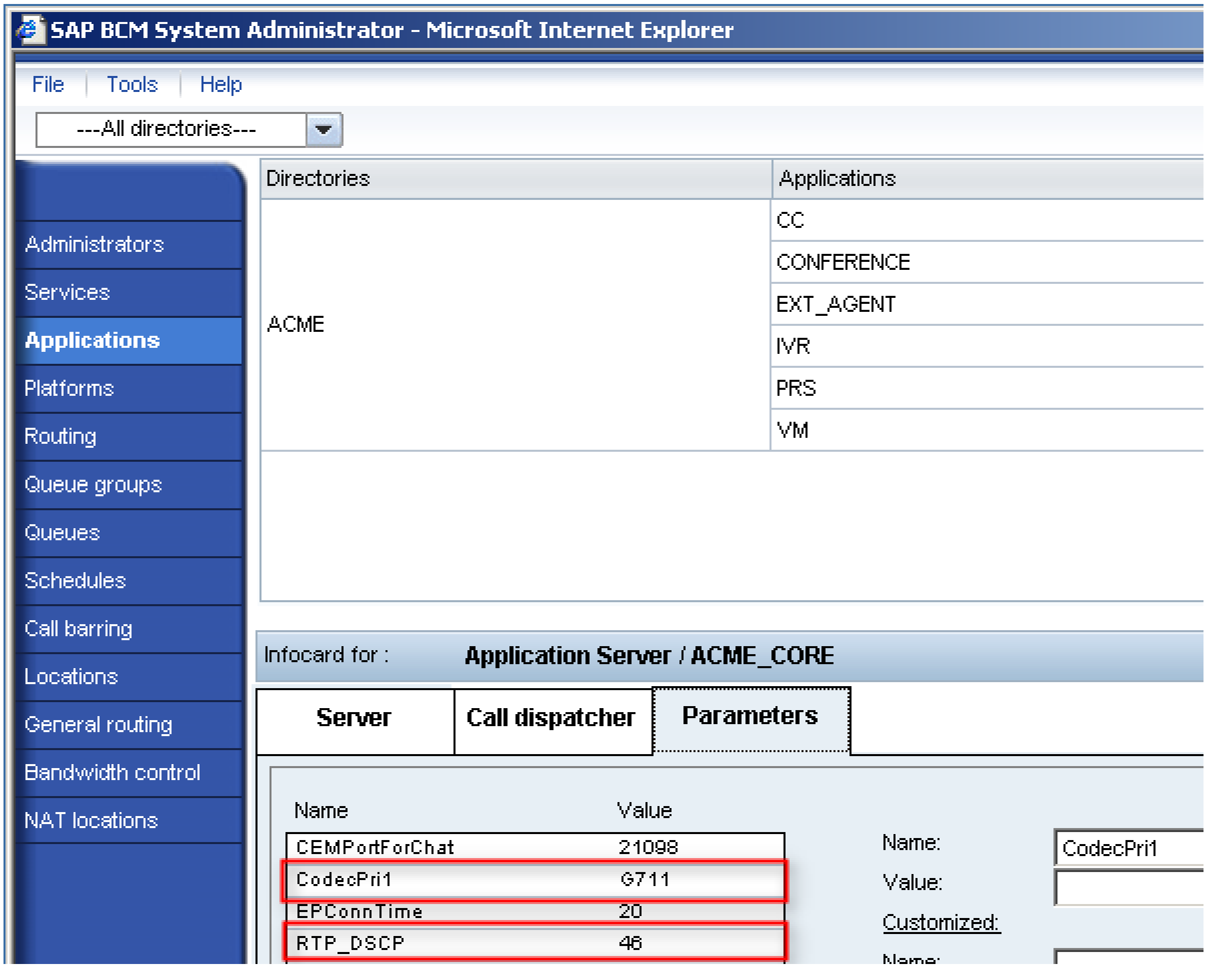
The expedited forwarding uses a DSCP name of EF, whose binary value is 101110 and decimal value of 46.

Figure 9: DSCP value



Note: Analyzing RTP package is possible to identify the parameters "Differentiated Services Field" with 10110 that correspond with 46 (Expedited Forwarding).

Figure 10: Adjusting DSCP value



Note: It's possible adjust the DSCP value in CEM Server (Call Dispatcher) using the parameter "RTP_DSCP" and the default Codec using the parameter "CodecPri1".

Related Content

BCM System Administrator – Administration Guide ver. 6.4

Cisco QOS Exam Certification Guide, 2nd Edition

Cisco Voice over IP (CVOICE), 3rd Edition

G.114 – ITU Recommendation G.114 – One-way Transmission Time

G.711 – ITU Recommendation G.711

G.729 – ITU Recommendation G.729

H.323 – ITU Recommendation H.323

RFC 1889 – RTP: A Transport Protocol for Real-Time Applications

RFC 2474 – Definition of the Differentiated Services Field (DS Field) in the IPv4 and IPv6 Headers

RFC 2475 – An Architecture for Differentiated Services

RFC 2597 – Assured Forwarding PHB Group

RFC 2598 – An Expedited Forwarding PHB

RFC 2543 – SIP: Session Initiation Protocol

RFC 3261 – SIP: Session Initiation Protocol

RFC 3465 – Session Initiation Protocol (SIP) Basic Call Flow Examples

RFC 3550 – RTP: A Transport Protocol for Real-Time Applications

Wireshark – Network protocol analyzer

For more information, visit the Customer Relationship Management homepage

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