

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.

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Created on: 23 March 2010

Author Bio



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Table of Contents

System Landscape	3
VoIP Signaling protocols.....	4
H.323 protocol stack	4
SIP protocol – Session Initiation Protocol	7
RTP and RTCP protocols.....	9
Codecs	10
VoIP bandwidth per call	11
Quality of Services – QoS.....	12
QoS tools	12
Differentiated Services – DSCP	12
Expedited Forwarding and DSCP Values	13
Related Content	15
Disclaimer and Liability Notice.....	16

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

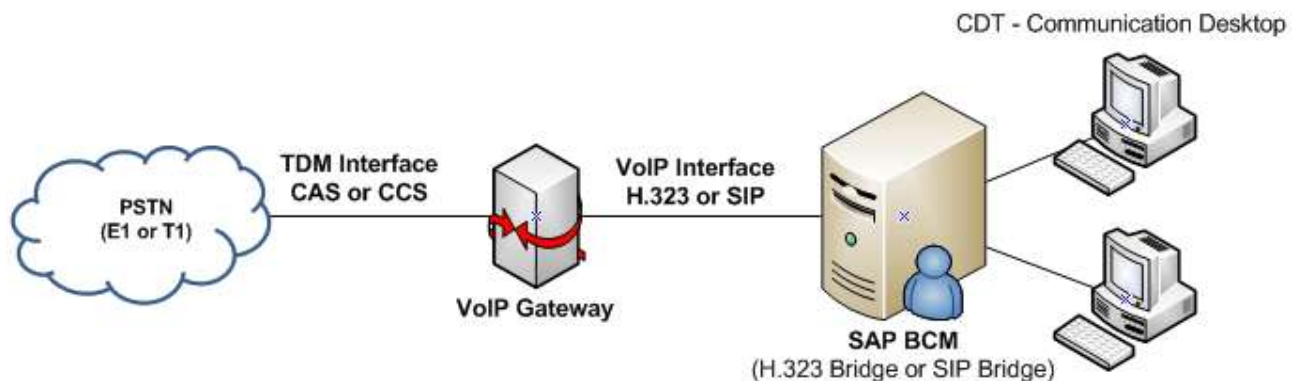
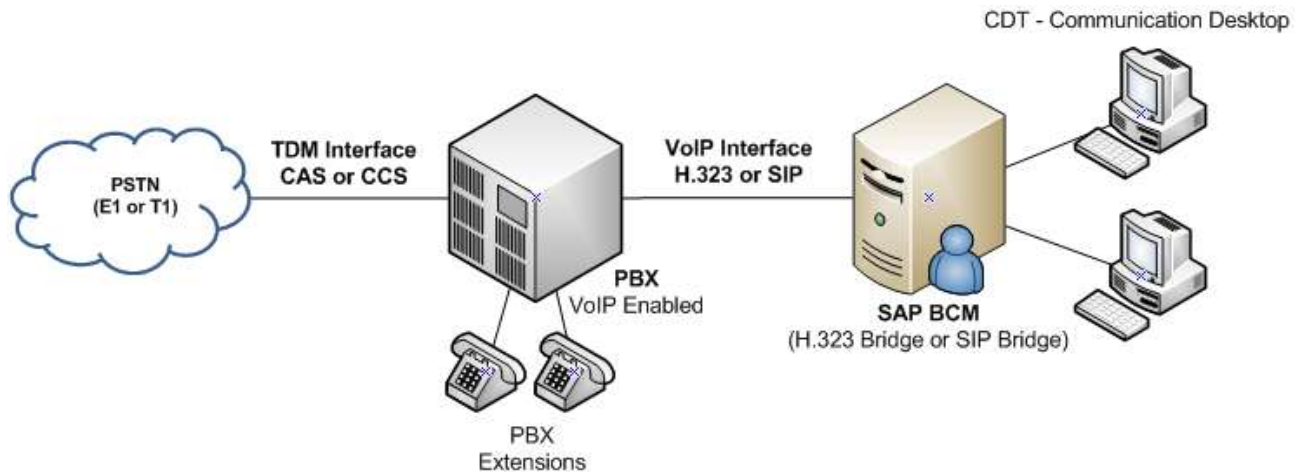


Figure 2: Using a PBX VoIP-enabled



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)

No. -	Time	Source	Destination	Protocol	Info
26	14.5	10.100.1.5	10.100.1.20	H.225.0	CS: setup OpenLogicalCha
27	14.5	10.100.1.20	10.100.1.5	H.225.0	CS: callProceeding
29	14.5	10.100.1.20	10.100.1.5	H.225.0	CS: alerting
55	20.7	10.100.1.20	10.100.1.5	H.225.0/H.245	CS: connect terminalCapa
57	20.7	10.100.1.5	10.100.1.20	H.225.0/H.245	CS: empty terminalCapabi
62	20.9	10.100.1.5	10.100.1.20	H.225.0/H.225	CS: empty masterSlaveDet
63	20.9	10.100.1.20	10.100.1.5	H.225.0/H.245	CS: empty masterSlaveDet
64	20.9	10.100.1.5	10.100.1.20	H.225.0/H.245	CS: empty masterSlaveDet
65	20.9	10.100.1.20	10.100.1.5	H.225.0/H.225	CS: empty terminalCapabi
66	20.9	10.100.1.5	10.100.1.20	H.225.0/H.245	CS: empty openLogicalCha
67	20.9	10.100.1.20	10.100.1.5	H.225.0/H.245	CS: empty openLogicalCha
68	20.9	10.100.1.5	10.100.1.20	H.225.0/H.245	CS: empty openLogicalCha
1250	29.3	10.100.1.5	10.100.1.20	H.225.0/H.245	CS: empty endSessionComm
1251	29.3	10.100.1.5	10.100.1.20	H.225.0	CS: releaseComplete
1253	29.3	10.100.1.20	10.100.1.5	H.225.0/H.245	CS: empty endSessionComm
1255	29.3	10.100.1.20	10.100.1.5	H.225.0	CS: releaseComplete


```

+ Frame 26 (576 bytes on wire, 576 bytes captured)
+ Ethernet II, Src: AlcatelB_53:ef:7c (00:80:9f:53:ef:7c), Dst: Dell_55:d5:
+ Internet Protocol, Src: 10.100.1.5 (10.100.1.5), Dst: 10.100.1.20 (10.100
+ Transmission Control Protocol, Src Port: 21080 (21080), Dst Port: h323hos
+ TPKT, Version: 3, Length: 510
- Q.931
  Protocol discriminator: Q.931
  Call reference value length: 2
  Call reference flag: Message sent from originating side
  Call reference value: 0035
  Message type: SETUP (0x05)
  Sending complete
+ Bearer capability
+ Progress indicator
+ Calling party number: '1180889597'
+ Called party number: '3275'
+ User-user
+ H.225.0 CS

```

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)

No. .	Time	Source	Destination	Protocol	Info
27	14.5	10.100.1.20	10.100.1.5	H.225.0	CS: callProceeding
29	14.5	10.100.1.20	10.100.1.5	H.225.0	CS: alerting
55	20.7	10.100.1.20	10.100.1.5	H.225.0/H.245	CS: connect terminalCapa
57	20.7	10.100.1.5	10.100.1.20	H.225.0/H.245	CS: empty terminalCapabi
62	20.9	10.100.1.5	10.100.1.20	H.225.0/H.225	CS: empty masterSlaveDet
63	20.9	10.100.1.20	10.100.1.5	H.225.0/H.245	CS: empty masterSlaveDet
64	20.9	10.100.1.5	10.100.1.20	H.225.0/H.245	CS: empty masterSlaveDet
65	20.9	10.100.1.20	10.100.1.5	H.225.0/H.225	CS: empty terminalCapabi
66	20.9	10.100.1.5	10.100.1.20	H.225.0/H.245	CS: empty openLogicalCha
67	20.9	10.100.1.20	10.100.1.5	H.225.0/H.245	CS: empty openLogicalCha
68	20.9	10.100.1.5	10.100.1.20	H.225.0/H.245	CS: empty openLogicalCha
1250	29.3	10.100.1.5	10.100.1.20	H.225.0/H.245	CS: empty endSessionComm
1251	29.3	10.100.1.5	10.100.1.20	H.225.0	CS: releaseComplete
1253	29.3	10.100.1.20	10.100.1.5	H.225.0/H.245	CS: empty endSessionComm
1255	29.3	10.100.1.20	10.100.1.5	H.225.0	CS: releaseComplete

- ⊕ Frame 66 (110 bytes on wire, 110 bytes captured)
- ⊕ Ethernet II, Src: AlcatelB_53:ef:7c (00:80:9f:53:ef:7c), Dst: Dell_55:d5:
- ⊕ Internet Protocol, Src: 10.100.1.5 (10.100.1.5), Dst: 10.100.1.20 (10.100.1.20)
- ⊕ Transmission Control Protocol, Src Port: 21080 (21080), Dst Port: h323hos
- ⊕ TPKT, Version: 3, Length: 44
- ⊕ Q.931
- ⊖ H.225.0 CS
 - ⊖ H323-UserInformation
 - ⊖ h323-uu-pdu
 - ⊕ h323-message-body: empty (8)
 - 1... h245Tunneling: True
 - ⊖ h245Control: 1 item
 - ⊖ Item 0
 - H245Control item: 20 octets
 - ⊖ H.245
 - ⊖ PDU Type: request (0)
 - ⊖ request: openLogicalChannel (3)
 - ⊖ openLogicalChannel
 - forwardLogicalChannelNumber: 288
 - ⊖ forwardLogicalChannelParameters
 - ⊖ dataType: audioData (3)
 - ⊖ audioData: g711Alaw64k (1)
 - g711Alaw64k: 20
 - ⊕ multiplexParameters: h2250LogicalChannelParameters (3)

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)

| No. - | Time | Source | Destination | Protocol | Info |
|-------|------|---------------|---------------|----------|--|
| 101 | 8.55 | 192.168.1.20 | 192.168.1.203 | SIP/SDP | Request: INVITE sip:1010@192.168.1.203 |
| 102 | 8.56 | 192.168.1.203 | 192.168.1.20 | SIP | Status: 100 Trying |
| 104 | 8.59 | 192.168.1.203 | 192.168.1.20 | SIP | Status: 180 Ringing |
| 121 | 11.0 | 192.168.1.203 | 192.168.1.20 | SIP/SDP | Status: 200 OK, with session description |
| 123 | 11.0 | 192.168.1.20 | 192.168.1.203 | SIP | Request: ACK sip:1010@192.168.1.203 |
| 2016 | 16.1 | 192.168.1.203 | 192.168.1.20 | SIP | Request: BYE sip:5008@192.168.1.203 |
| 2043 | 16.3 | 192.168.1.20 | 192.168.1.203 | SIP | Status: 200 OK |


```

+ Frame 101 (1040 bytes on wire, 1040 bytes captured)
  Ethernet II, Src: vmware_fe:2d:de (00:0c:29:fe:2d:de), Dst: vmware_46:63:
  Internet Protocol, Src: 192.168.1.20 (192.168.1.20), Dst: 192.168.1.203 (
  Transmission Control Protocol, Src Port: 37678 (37678), Dst Port: sip (50
  Session Initiation Protocol
    Request-Line: INVITE sip:1010@192.168.1.203:5060 SIP/2.0
    Message Header
      Via: SIP/2.0/TCP 192.168.1.20:5060;branch=z9hg4bk7c13b05eb0
      Remote-Party-ID: <sip:5008@192.168.1.20>;party=calling;screen=yes;pri
      From: <sip:5008@192.168.1.20>;tag=1f8da27f-d49f-4313-bb44-9b9d7b02e79
      To: <sip:1010@192.168.1.203>
      Date: Mon, 27 Apr 2009 03:13:06 GMT
      Call-ID: 52685280-9f5122c2-3b-1401a8c0@192.168.1.20
      Supported: timer,replaces
      Min-SE: 1800
      User-Agent: Cisco-CCM6.0
      Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER,
    CSeq: 101 INVITE
    Contact: <sip:5008@192.168.1.20:5060;transport=tcp>
      Expires: 180
      Allow-Events: presence, kpm1
      Session-Expires: 1800
      Max-Forwards: 70
      Content-Type: application/sdp
      Content-Length: 212
    Message Body
  
```

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)

| No. - | Time | Source | Destination | Protocol | Info |
|-------|------|---------------|---------------|----------|--|
| 101 | 8.55 | 192.168.1.20 | 192.168.1.203 | SIP/SDP | Request: INVITE sip:1010@192.168.1.203 |
| 102 | 8.56 | 192.168.1.203 | 192.168.1.20 | SIP | Status: 100 Trying |
| 104 | 8.59 | 192.168.1.203 | 192.168.1.20 | SIP | Status: 180 Ringing |
| 121 | 11.0 | 192.168.1.203 | 192.168.1.20 | SIP/SDP | Status: 200 OK, with session |
| 123 | 11.0 | 192.168.1.20 | 192.168.1.203 | SIP | Request: ACK sip:1010@192.168.1.203 |
| 2016 | 16.1 | 192.168.1.203 | 192.168.1.20 | SIP | Request: BYE sip:5008@192.168.1.203 |
| 2043 | 16.3 | 192.168.1.20 | 192.168.1.203 | SIP | Status: 200 OK |


```

+ Frame 101 (1040 bytes on wire, 1040 bytes captured)
+ Ethernet II, Src: Vmware_fe:2d:de (00:0c:29:fe:2d:de), Dst: Vmware_46:63:
+ Internet Protocol, src: 192.168.1.20 (192.168.1.20), Dst: 192.168.1.203 (
+ Transmission Control Protocol, Src Port: 37678 (37678), Dst Port: sip (50
+ Session Initiation Protocol
+ Request-Line: INVITE sip:1010@192.168.1.203:5060 SIP/2.0
+ Message Header
+ Message Body
+ Session Description Protocol
  Session Description Protocol Version (v): 0
  + Owner/Creator, Session Id (o): CiscoSystemsCCM-SIP 2000 1 IN IP4 19
    Session Name (s): SIP Call
  + Connection Information (c): IN IP4 192.168.1.20
  + Time Description, active time (t): 0 0
  + Media Description, name and address (m): audio 24588 RTP/AVP 0 101
  + Media Attribute (a): rtpmap:0 PCMU/8000
  + Media Attribute (a):ptime:20
  + Media Attribute (a): rtpmap:101 telephone-event/8000
  + Media Attribute (a): fmp:101 0-15

```

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

| No. . | Time | Source | Destination | Protocol | Info |
|---|------|------------|-------------|----------|--------------------------------|
| 59136 | 28.1 | 10.100.1.5 | 192.168.13. | RTCP | Receiver Report Source descrip |
| + Frame 59137 (126 bytes on wire, 126 bytes captured) | | | | | |
| + Ethernet II, Src: HewlettP_36:ee:e8 (00:23:7d:36:ee:e8), Dst: Cisco_c6:bc | | | | | |
| + Internet Protocol, Src: 192.168.13.34 (192.168.13.34), Dst: 10.100.1.5 (1 | | | | | |
| - Internet Control Message Protocol | | | | | |
| Type: 3 (Destination unreachable) | | | | | |
| Code: 3 (Port unreachable) | | | | | |
| checksum: 0xa14b [correct] | | | | | |
| + Internet Protocol, Src: 10.100.1.5 (10.100.1.5), Dst: 192.168.13.34 (19 | | | | | |
| + User Datagram Protocol, Src Port: 32553 (32553), Dst Port: 8047 (8047) | | | | | |
| - Real-time Transport Control Protocol (Receiver Report) | | | | | |
| + [Stream setup by H245 (frame 36893)] | | | | | |
| 10.. = Version: RFC 1889 version (2) | | | | | |
| ..0. = Padding: False | | | | | |
| ...0 0001 = Reception report count: 1 | | | | | |
| Packet type: Receiver Report (201) | | | | | |
| Length: 7 (32 bytes) | | | | | |
| Sender SSRC: 0xa7624cad (2808237229) | | | | | |
| - Source 1 | | | | | |
| Identifier: 0xc2e2fd78 (3269655928) | | | | | |
| - SSRC contents | | | | | |
| Fraction lost: 0 / 256 | | | | | |
| Cumulative number of packets lost: 0 | | | | | |
| + Extended highest sequence number received: 3827 | | | | | |
| Interarrival jitter: 38 | | | | | |
| Last SR timestamp: 0 (0x00000000) | | | | | |
| Delay since last SR timestamp: 0 (0 milliseconds) | | | | | |
| + Real-time Transport Control Protocol (Source description) | | | | | |

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.

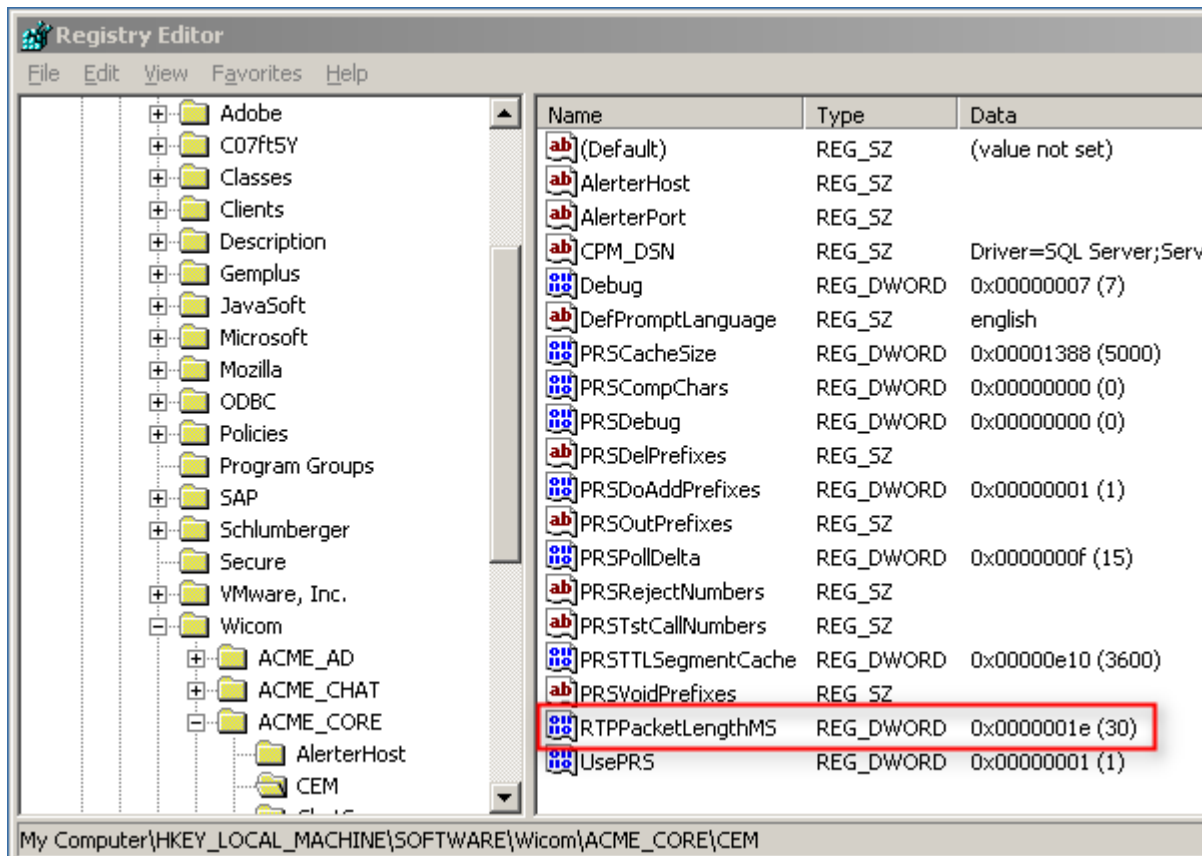
Table 1: Packets to transmit on second of conversation.

| 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:

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

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

$$\text{Total Bandwidth (Kbps)} = 31.2 \text{ Kbps (per call using G.729 codec)}$$

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

| No. . | Time | Source | Destination | Protocol | Info |
|---|------|------------|-------------|----------|--------------------------|
| 84 | 21.1 | 10.100.1.3 | 10.100.1.20 | RTP | PT=ITU-T G.711 PCMA, SSR |
| + Frame 88 (214 bytes on wire, 214 bytes captured) | | | | | |
| + Ethernet II, Src: AlcatelB_53:ef:7b (00:80:9f:53:ef:7b), Dst: Dell_55:d5: | | | | | |
| + Internet Protocol, Src: 10.100.1.3 (10.100.1.3), Dst: 10.100.1.20 (10.100.1.20) | | | | | |
| Version: 4 | | | | | |
| Header length: 20 bytes | | | | | |
| + Differentiated Services Field: 0xb8 (DSCP 0x2e: Expedited Forwarding; ECN: 0) | | | | | |
| 1011 10.. = Differentiated Services Codepoint: Expedited Forwarding (EF) | | | | | |
|0. = ECN-Capable Transport (ECT): 0 | | | | | |
|0 = ECN-CE: 0 | | | | | |
| Total Length: 200 | | | | | |
| Identification: 0x0000 (0) | | | | | |
| + Flags: 0x04 (Don't Fragment) | | | | | |
| Fragment offset: 0 | | | | | |
| Time to live: 64 | | | | | |
| Protocol: UDP (0x11) | | | | | |
| + Header checksum: 0x228f [correct] | | | | | |
| Source: 10.100.1.3 (10.100.1.3) | | | | | |
| Destination: 10.100.1.20 (10.100.1.20) | | | | | |
| + User Datagram Protocol, Src Port: 32544 (32544), Dst Port: 1m-perfworks (32544) | | | | | |
| + Real-Time Transport Protocol | | | | | |

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

The screenshot shows the SAP BCM System Administrator interface. The left sidebar contains a navigation menu with 'Applications' selected. The main content area displays a table of applications for the 'ACME' directory, including CC, CONFERENCE, EXT_AGENT, IVR, PRS, and VM. Below this, the 'Infocard for: Application Server / ACME_CORE' is shown. The 'Parameters' tab is active, displaying a table of parameters:

| Name | Value |
|----------------|-------|
| CEMPortForChat | 21098 |
| CodecPri1 | G711 |
| EPConnTime | 20 |
| RTP_DSCP | 46 |

Below the table, there are input fields for 'Name:' (containing 'CodecPri1'), 'Value:', and 'Customized:'.

Note: It's possible to 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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