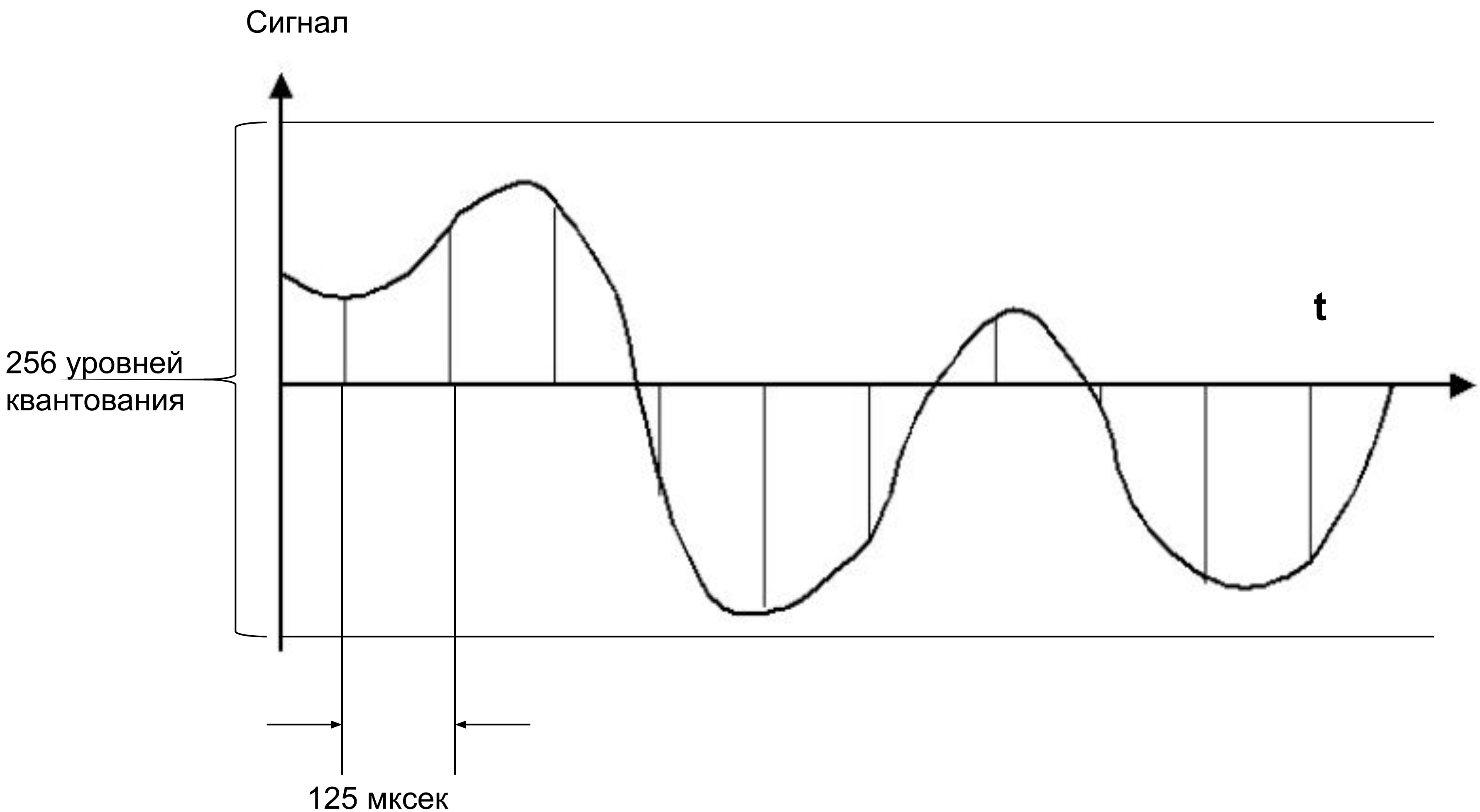
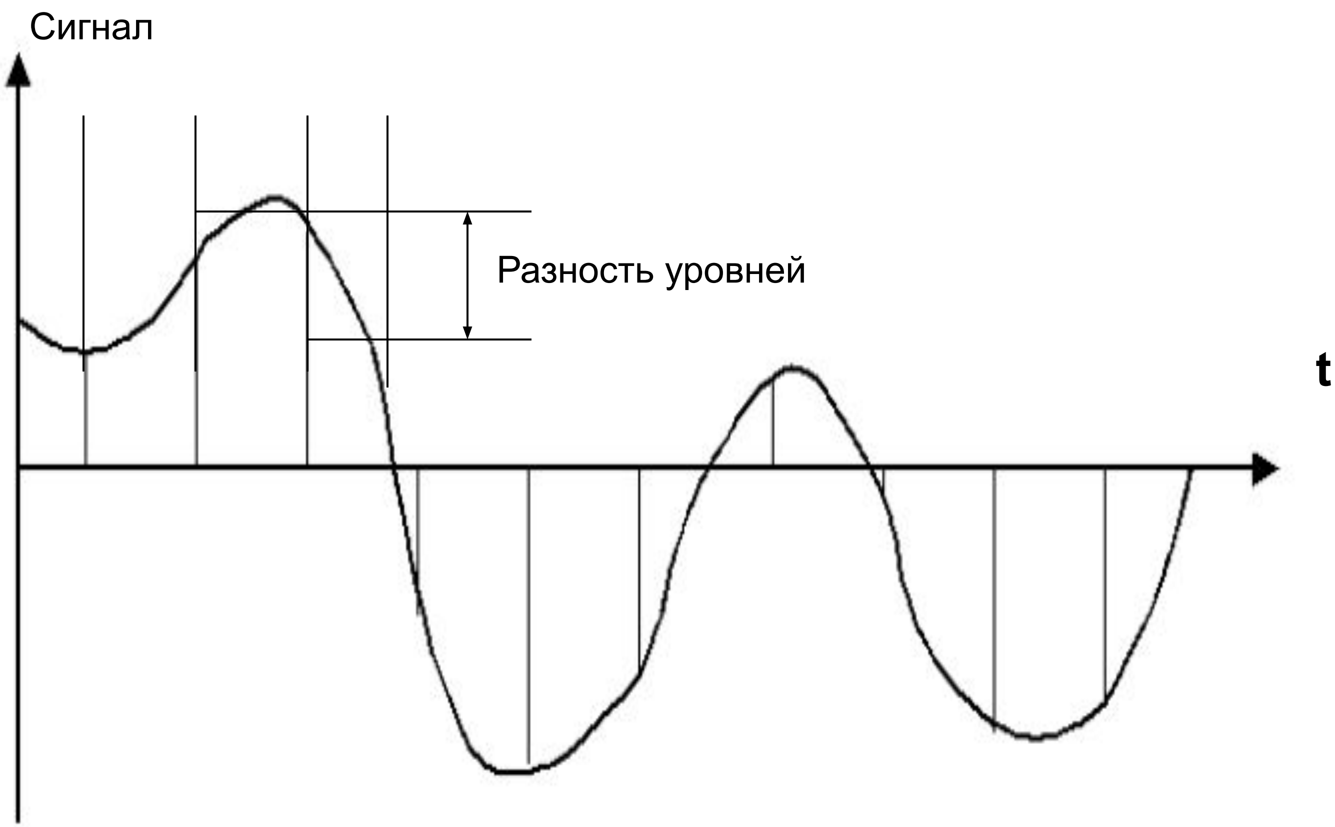
Технология VoIP

Профессор В.Ю.Деарт

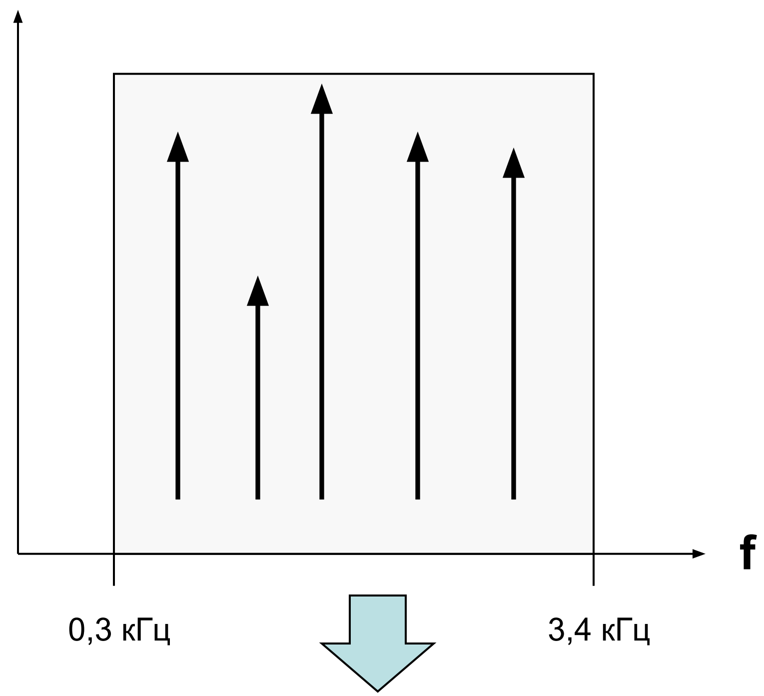
Импульсно-кодовая модуляция.



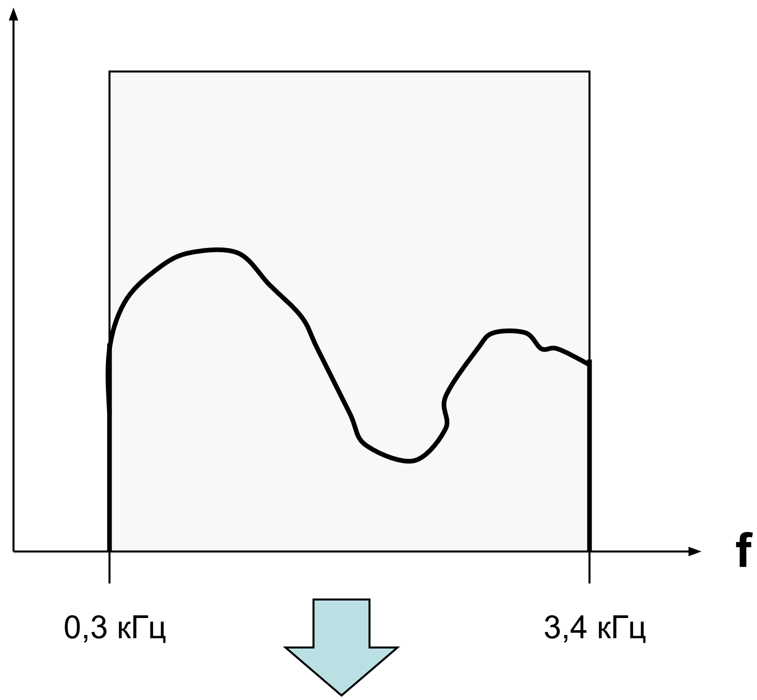
Адаптивная импульсно- кодовая модуляция.



Принципы речевого кодирования GSM



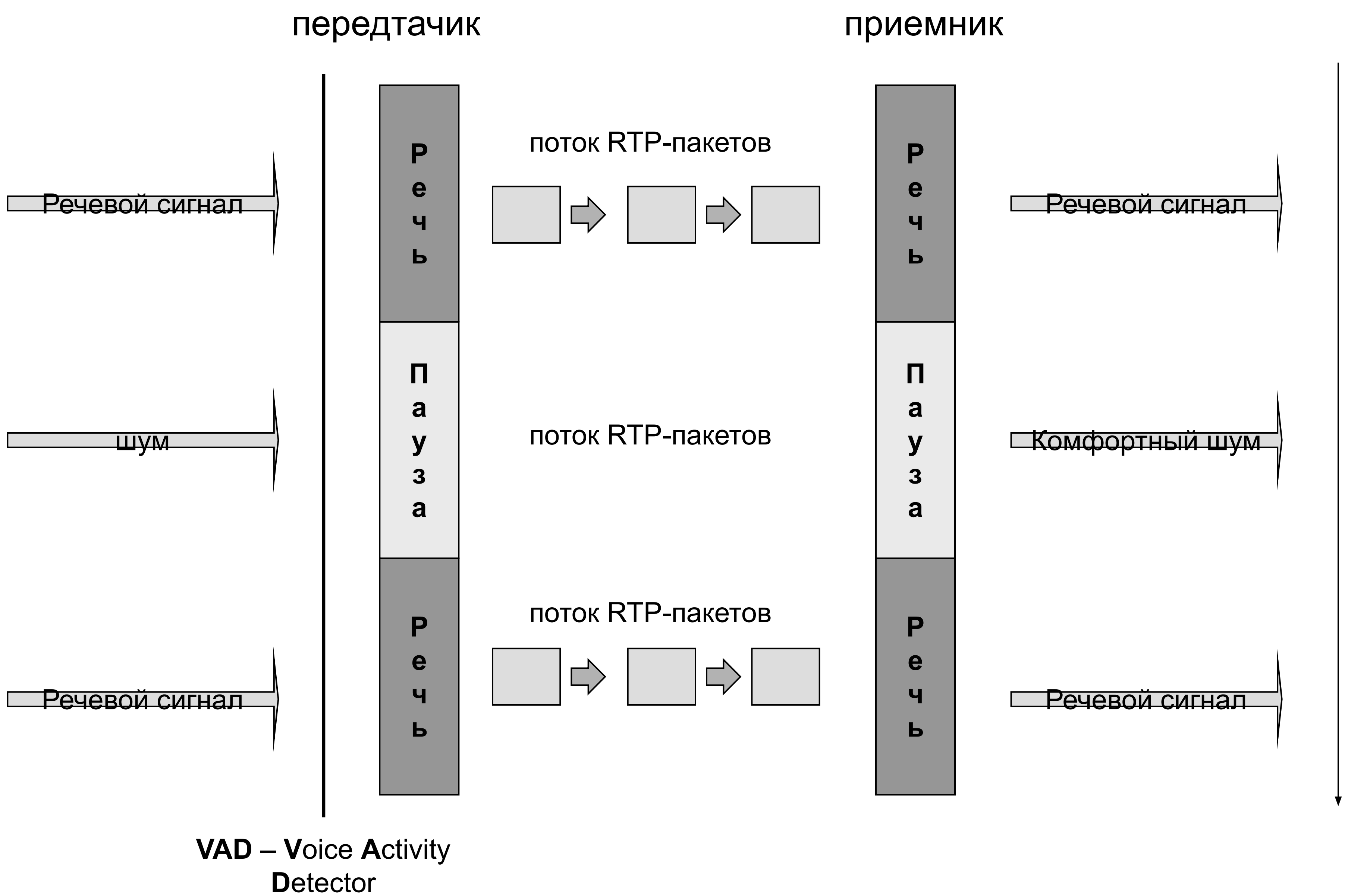
Каждый гласный звук представляется в виде набора гармонических сигна-лов



Каждый согласный звук представ-ляется в виде шумоподобного сиг-нала

AMR - Adaptive Multirate - адаптивный многоскоростной кодек

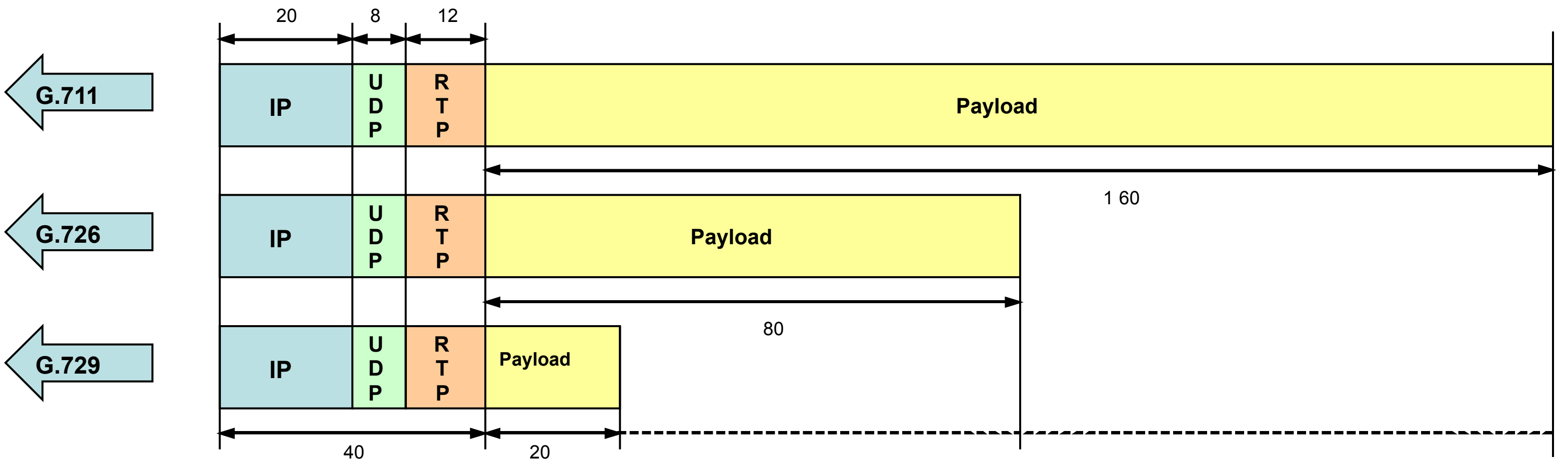
Использование речевых пауз



Стандарты передачи речевого сигнала.

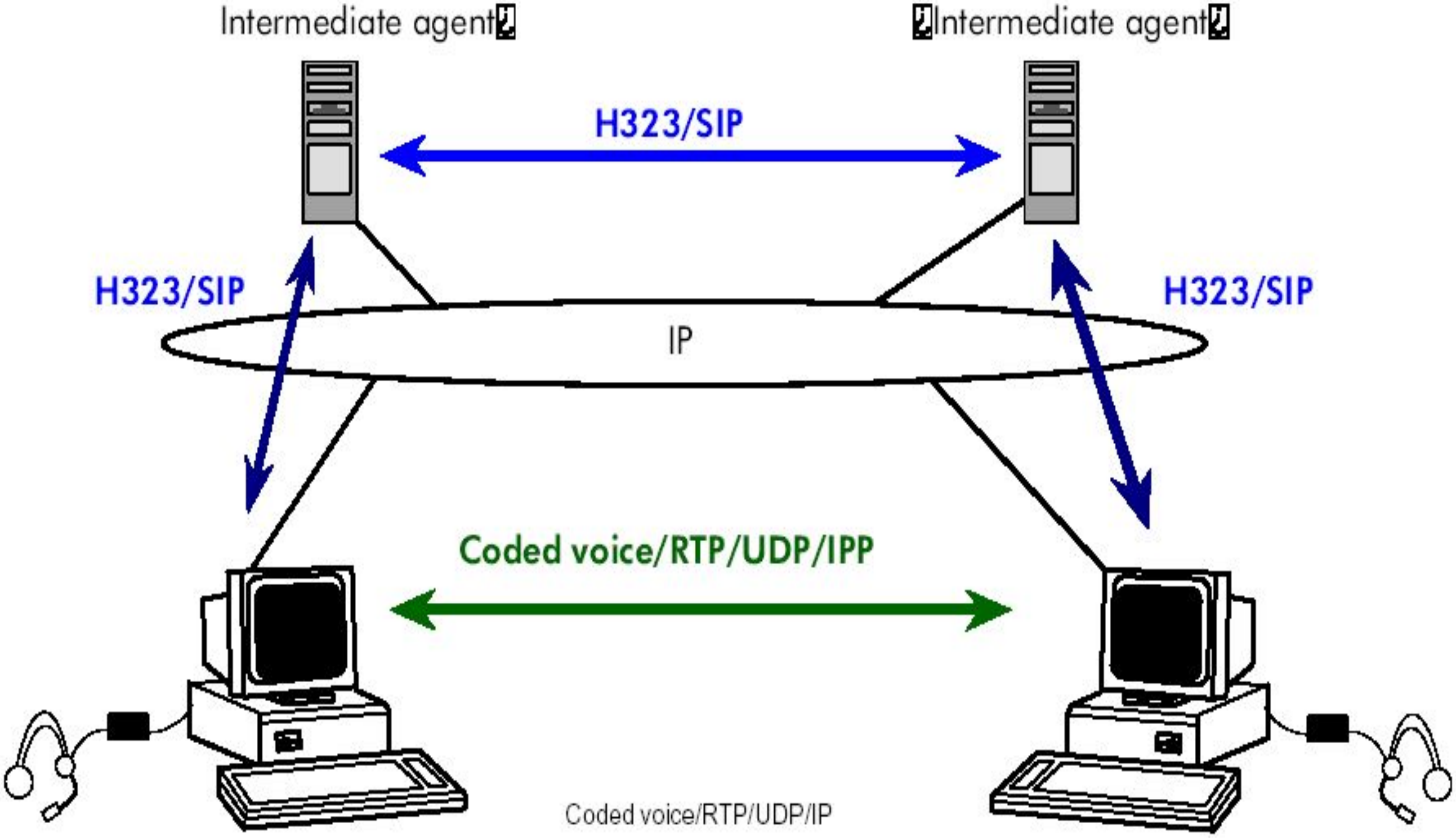
Стандарт	Скорость цифрового потока кбит/с	Задержка	Использование речевых пауз
G.711	64	125 мкс	-
G.726	32	125 мкс	-
AMR	12,2	20 мс	+
G.729	8	15 мс	+

Пакетизация речи в различных кодеках

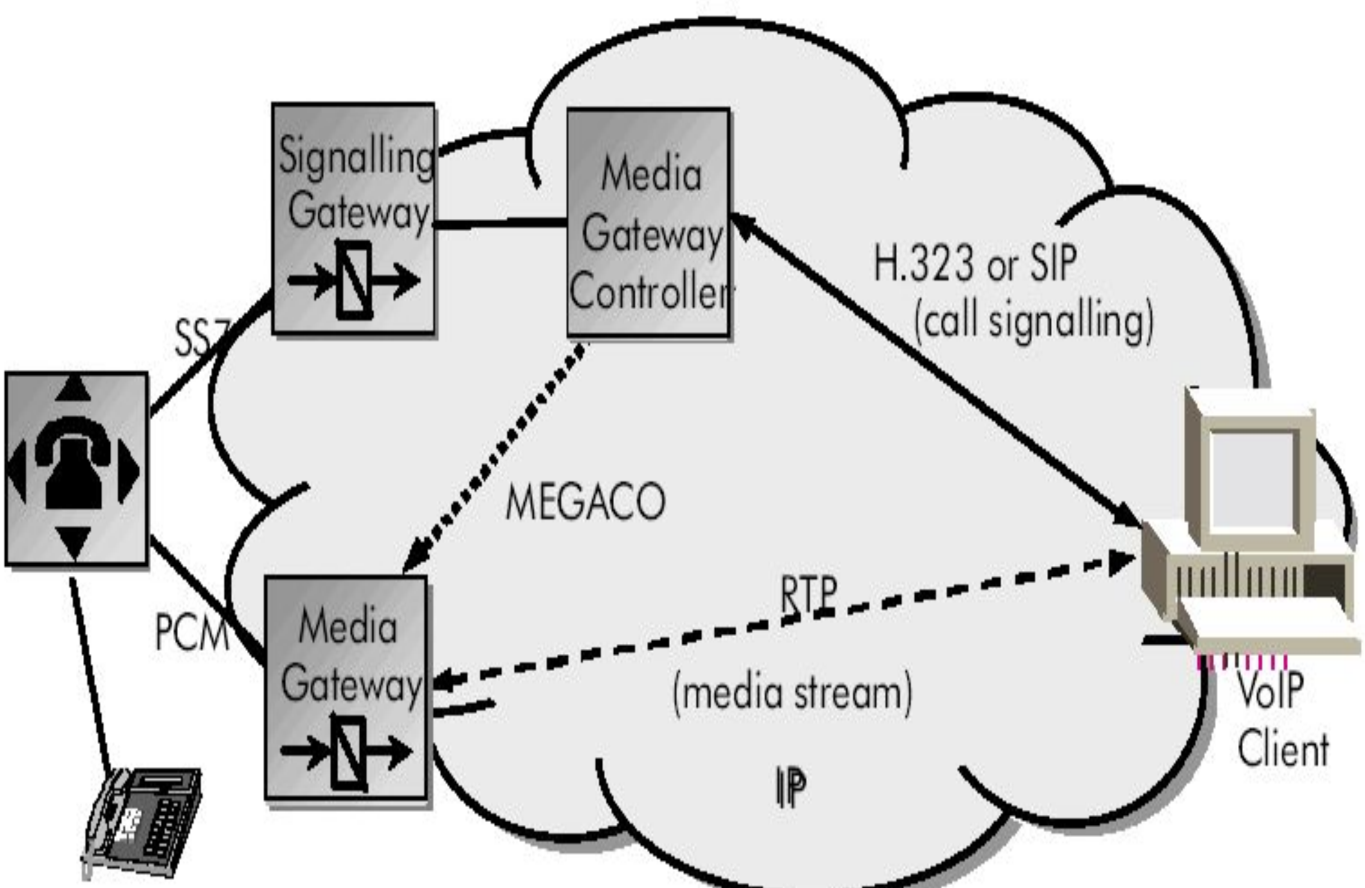


Кодек	Пакет С заголовком байт	Payload Байт	Передача речевого сигнала бит /с	Передача IP - пакетов бит /с
G.711	200	160	64	80
G.726	120	80	32	48
G.729	6 0	20	8	24

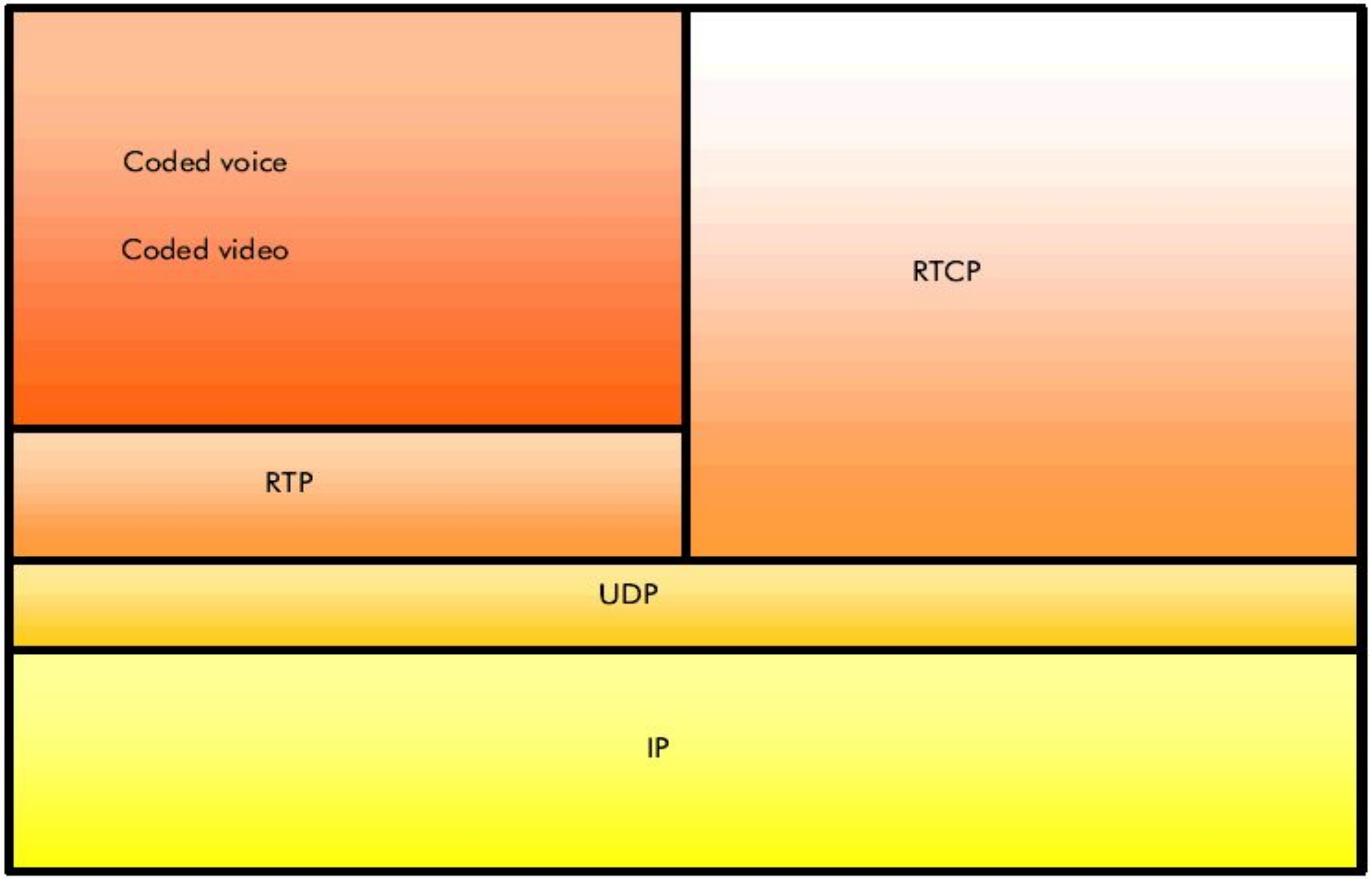
Структура протоколов VoIP



Протоколы взаимодействия



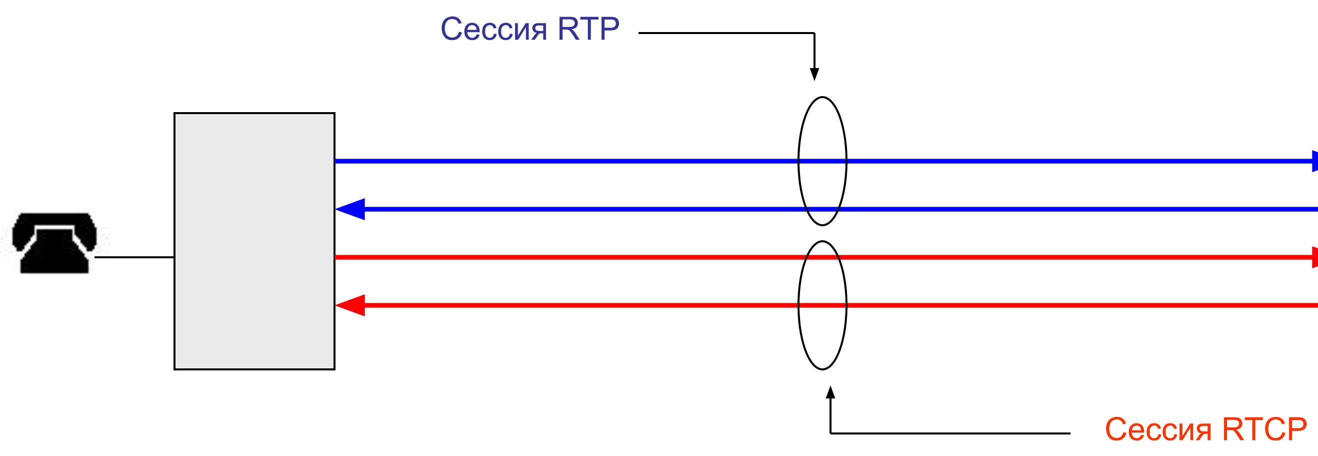
Структура стека протоколов передачи речи

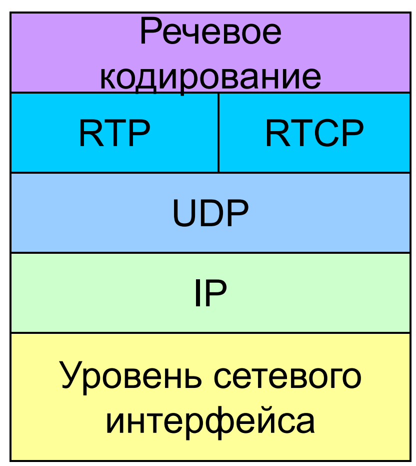


Список протоколов передачи речи и видео

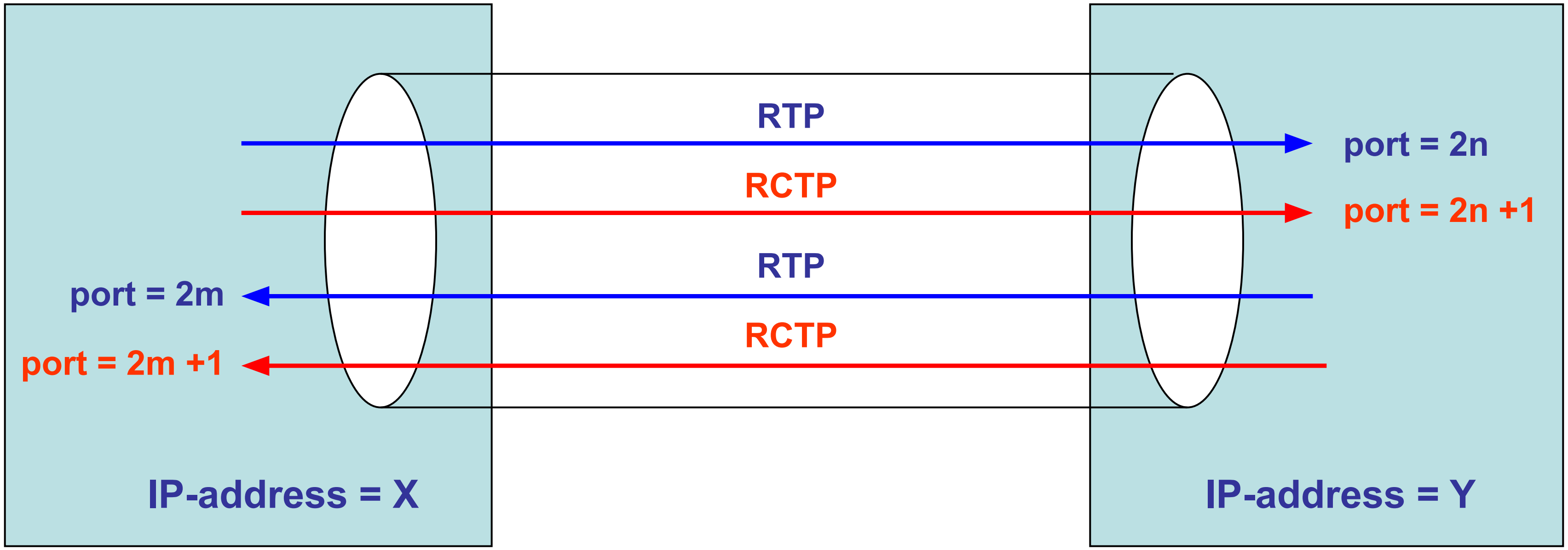
- **RTP – Real – time Transport Protocol (RFC 1889)**
- **RTCP – Real – time Transport Control Protocol**
- **UDP – User Datagram Protocol**
- **IP – Internet Protocol**

Протоколы RTP и RTCP.

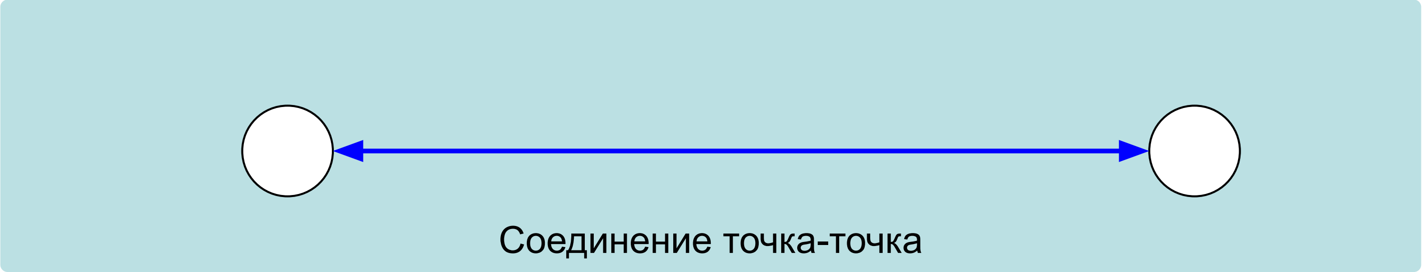


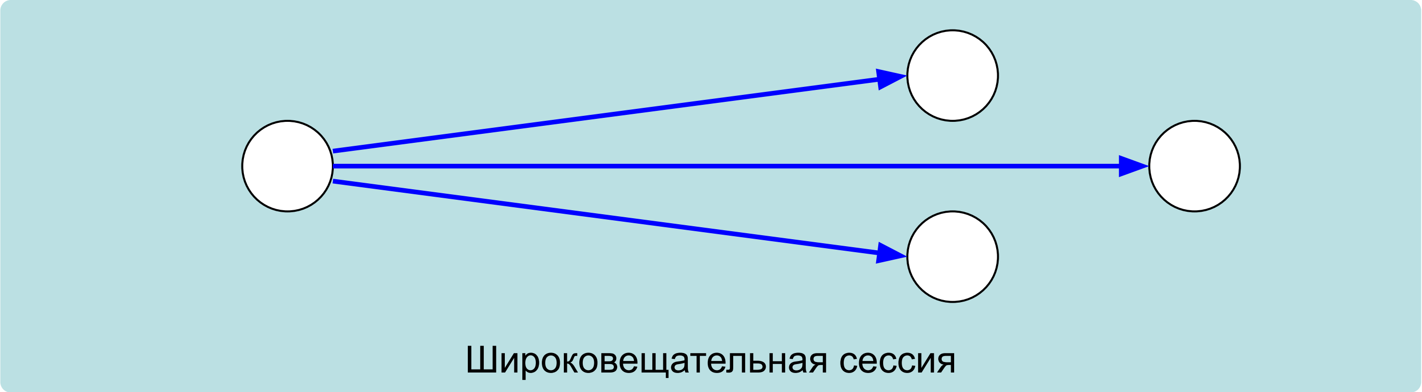


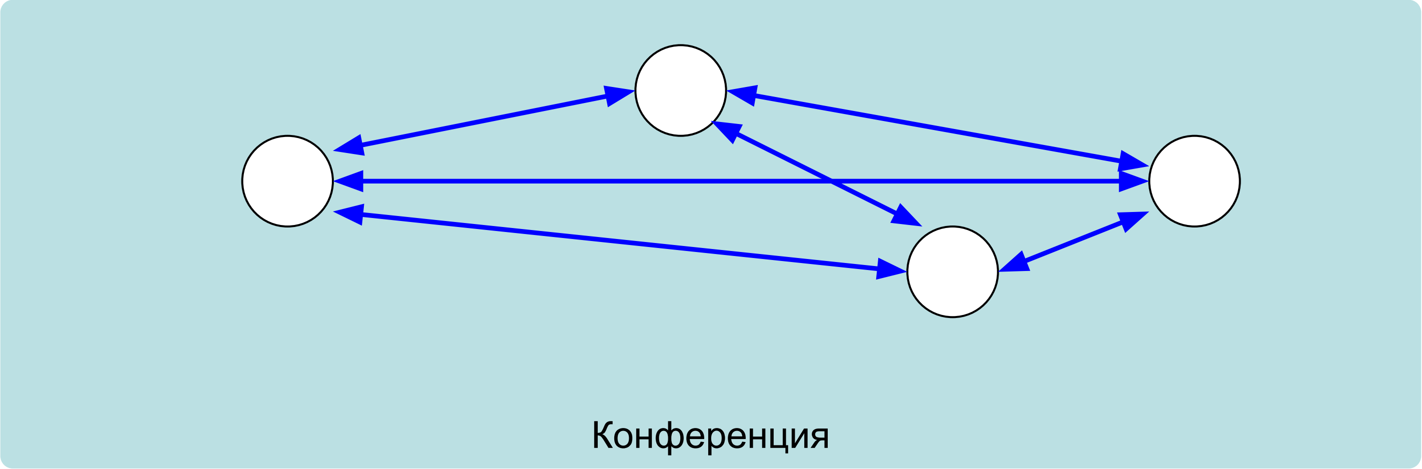
Номера портов RTP и RTCP.



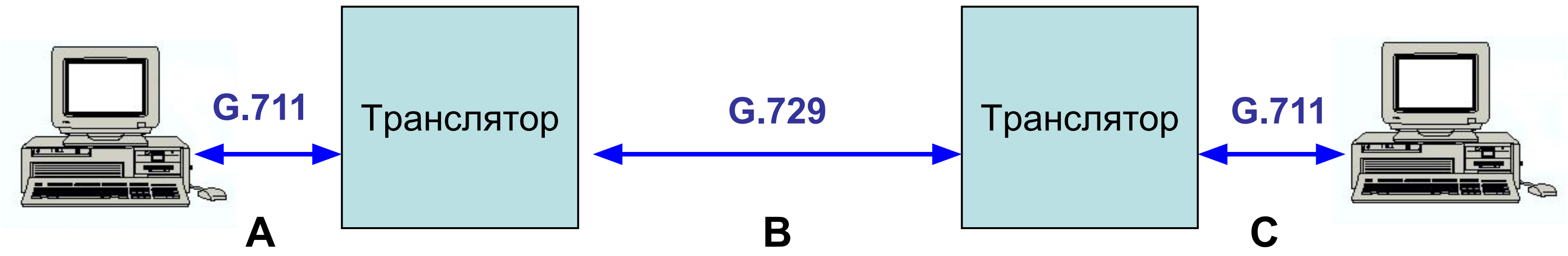
Конфигурации RTP-сессии.

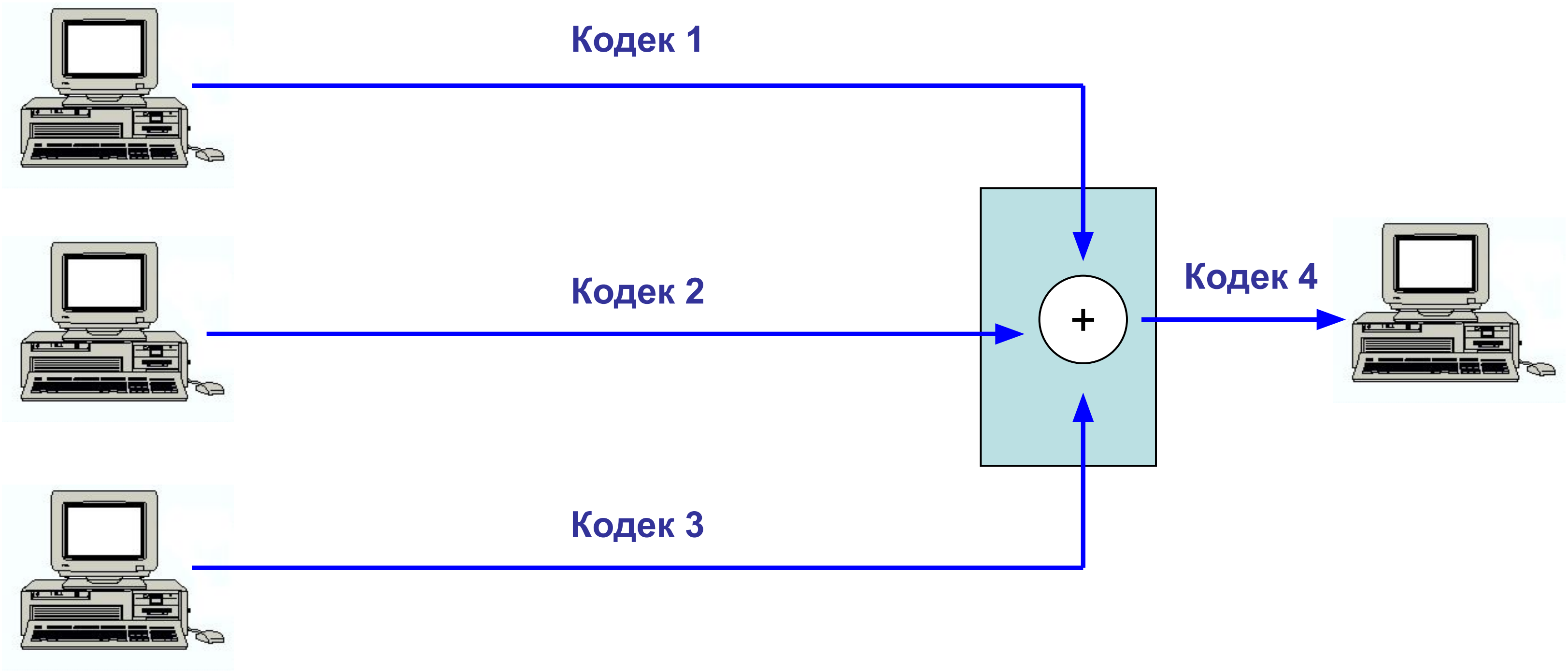




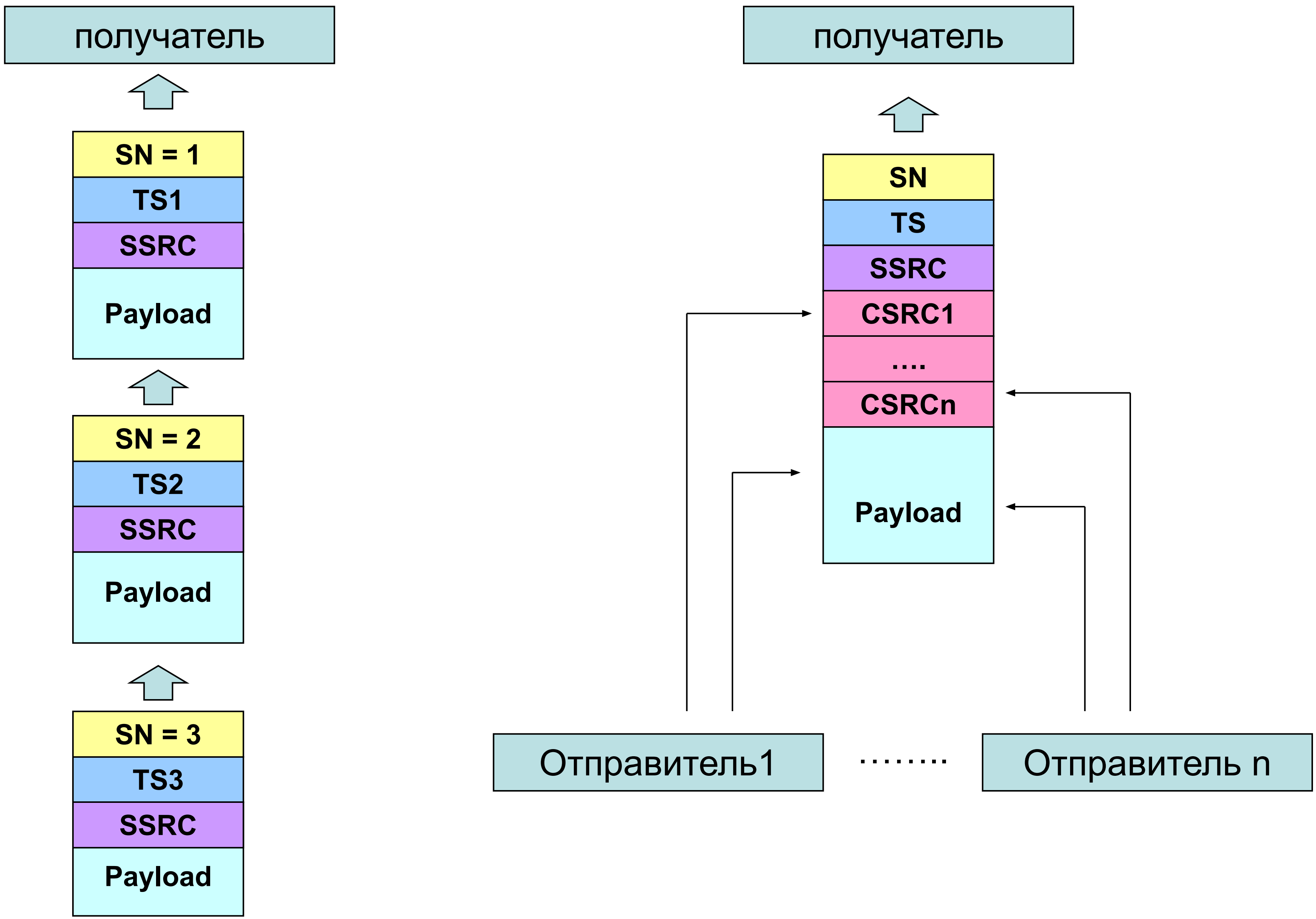


Трансляторы и смесители (1)

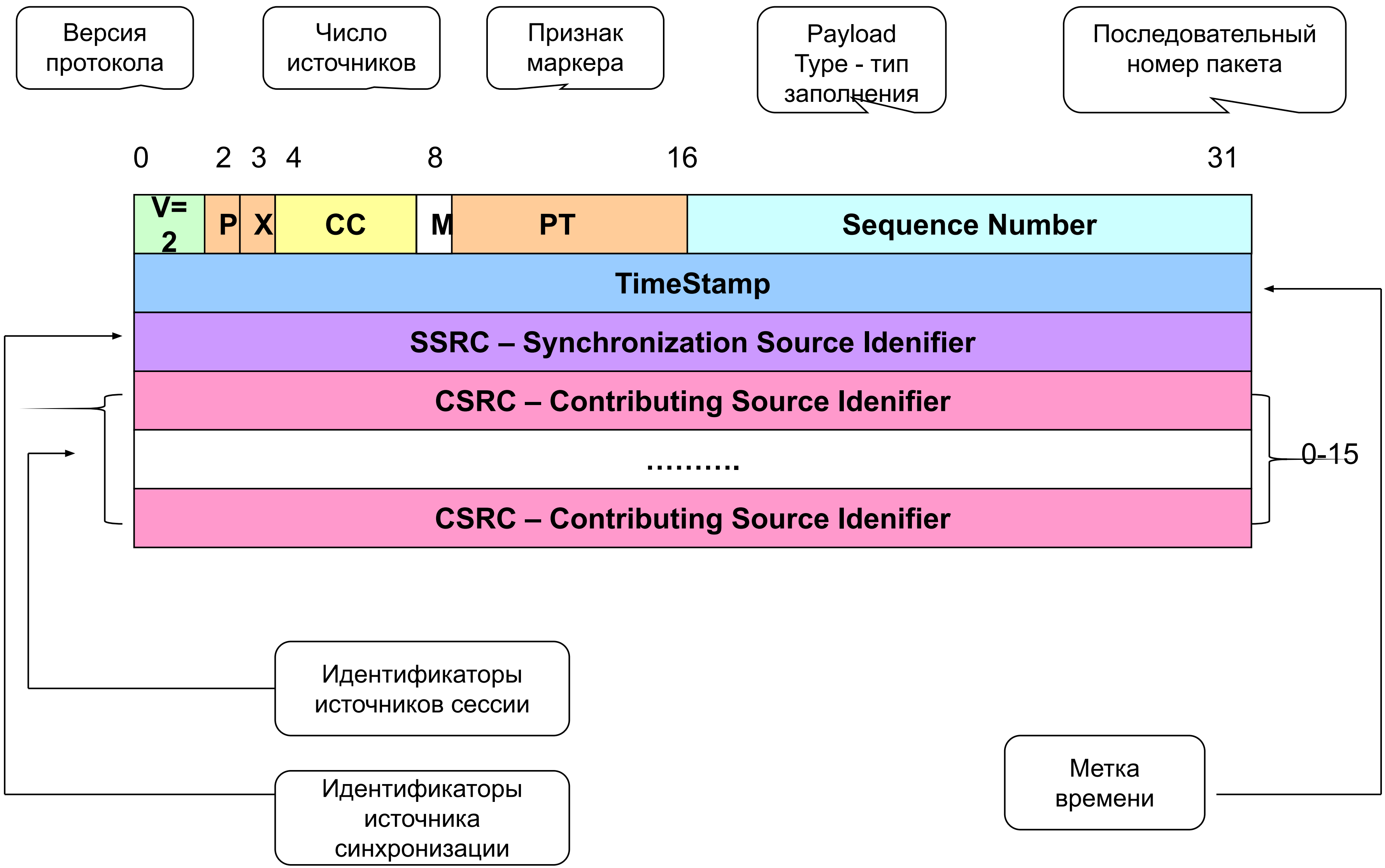




Служебная информация RTP-пакетов.



Формат заголовка RTP - пакета.



Назначение полей заголовка пакета RTP(1)

- **V**: two bits are reserved for the **RTP version**. At this moment version 2 is used (1 was a IETF draft version).
- **P**: If the **padding bit** is set (1), the packet contains one or more octets at the end which are not part of the payload (filler bits added for alignment purposes). The last octet of the payload indicates more precisely how many padding octets have been appended to the original payload (this is necessary since there is no length field available to specify the actual payload size)
- **X**: An **extension bit** indicates the presence of extensions after the eventual CSRCs of the fixed header.

  The purpose of the header extension is to allow implementors to experiment with features that are not part of the standard implementation.
- **CC**: The **CSRC count** (4 bits) contains the number of CSRC identifiers that follow the fixed header.
- **M**: The **marker bit** is intended to allow significant events such as frame boundaries to be marked in the packet stream.

  The use of the marker bit is implementation specific. A typical implementation is to mark the beginning of a speech burst if silence suppression is used. G.723.1 is an example of a codec that uses this technique.
- **PT**: The **Payload Type** field (7bits) identifies the format of the RTP payload.

Кодирование поля тип нагрузки (payload type)

0	G.711 mu-law PCM audio	18	G.729 audio
1	1016 audio	19-22	unassigned audio
2	G.721	23	unassigned audio
3	GSM 6.10 audio	24	unassigned audio
4	G.723.1 audio	25	Ce1B video
5	DV14 audio (8kHz)	26	JPEG video
6	DV14 audio (16kHz)	27	unassigned audio
7	LPC audio	28	nv video
8	G.711 A-law PCM audio	29-30	unassigned audio
9	G.722 audio (16kHz)	31	H.261 video
10	L16 audio (stereo)	32	MPV video
11	L16 audio (mono)	33	MP2T video
12	G.723	34	H.263 video
13	CN (comfort noise level)	35-71	unassigned
14	MPA audio	72-76	reserved
15	G.728 audio	77	RED audio
16	DV14 audio (11.025 kHz)	78-95	unassigned
17	DV14 audio (22.050 kHz)	96-127	dynamic

Назначение полей заголовка пакета RTP(2)

- The **Sequence number** (16 bits) increments by one for each RTP data packet sent. It may be used by the receiver to detect packet loss and to restore packet sequence. The initial value of the sequence number is random.
- The sampling clock inserts a **Timestamp** (32 bits), not the system clock. Therefore the timestamp in the RTP packet gives a relative timing not an absolute (network) timing.

  For video, the RTP timestamp is the tick count of the display time of the first frame encoded in the packet payload. For audio, the RTP timestamp is the tick count when the first audio sample contained in the payload was sampled. The clock is initialized with a random value.
- The **SSRC field identifier** (32 bits) is already discussed in previous paragraphs. The identifier number is chosen at random.
- **CSRC** (0 to 15 times 32 bits): A maximum of 15 contributing sources can be identified. More CSRCs can be used, but they can not be listed in the packet.

  <u>Remark</u>: if no mixer is used, the CC will be zero and no extra CSRC fields will be inserted.

Пример RTP – пакета (трассировка)

Frame 87 (214 bytes on wire, 214 bytes captured)
Ethernet II, Src: 00:a0:c5:5d:94:2c, Dst: 00:03:2f:09:ae:2f
Internet Protocol, Src Addr: 217.10.67.6 (217.10.67.6),
Dst Addr: 192.168.1.42 (192.168.1.42)
User Datagram Protocol, Src Port: 17130 (17130), **Dst Port:** irdmi (8000))
Real-Time Transport Protocol

```
10.. .... = Version: RFC 1889 Version (2)
..0. .... = Padding: False
...0 .... = Extension: False
.... 0000 = Contributing source identifiers count: 0
0... .... = Marker: False
.000 1000 = Payload type: ITU-T G.711 PCMA (8)
```

Sequence number: 1445
Timestamp: 320
Synchronization Source identifier: 1280623826
Payload: AAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAA...

Трасса передачи RTP пакетов(Ethereal)

- No. Time Source Destination Protocol Info
- 61 1.11128 192.168.0.183 192.168.0.187 RTP Payload Type=ITU-T G.711PCMA
- SSRC=33071136, Seq=22054, Time=0, Mark

- 62 1.111372 192.168.0.183 192.168.0.187 RTP Payload Type=ITU-T G.711PCMA
- SSRC=33071136, Seq=22055, Time=160, Mark

- 63 1.12588 192.168.0.187 192.168.0.183 RTP Payload Type=ITU-T G.711PCMA
- SSRC=435983184, Seq=7840, Time=0, Mark

- 64 1.126117 192.168.0.187 192.168.0.183 RTP Payload Type=ITU-T G.711PCMA
- SSRC=435983184, Seq=7841, Time=160, Mark

- 65 1.160160 192.168.0.183 192.168.0.187 RTP Payload Type=ITU-T G.711PCMA
- SSRC=33071136, Seq=22056, Time=320, Mark

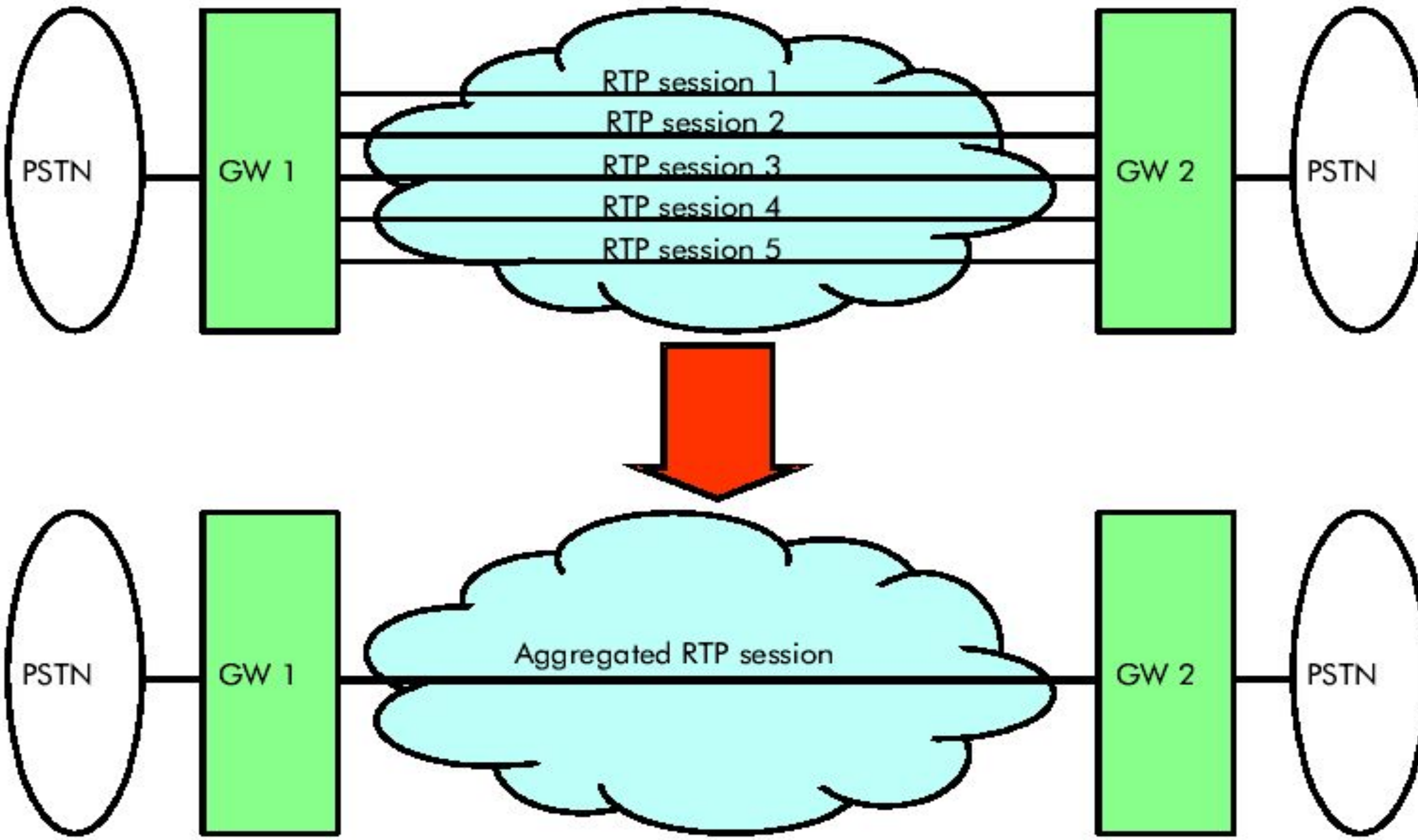
Размеры заголовков стека RTP/UDP/IP/Ethernet

Overheads are as follows (assume RTP/UDP/IP/Ethernet):

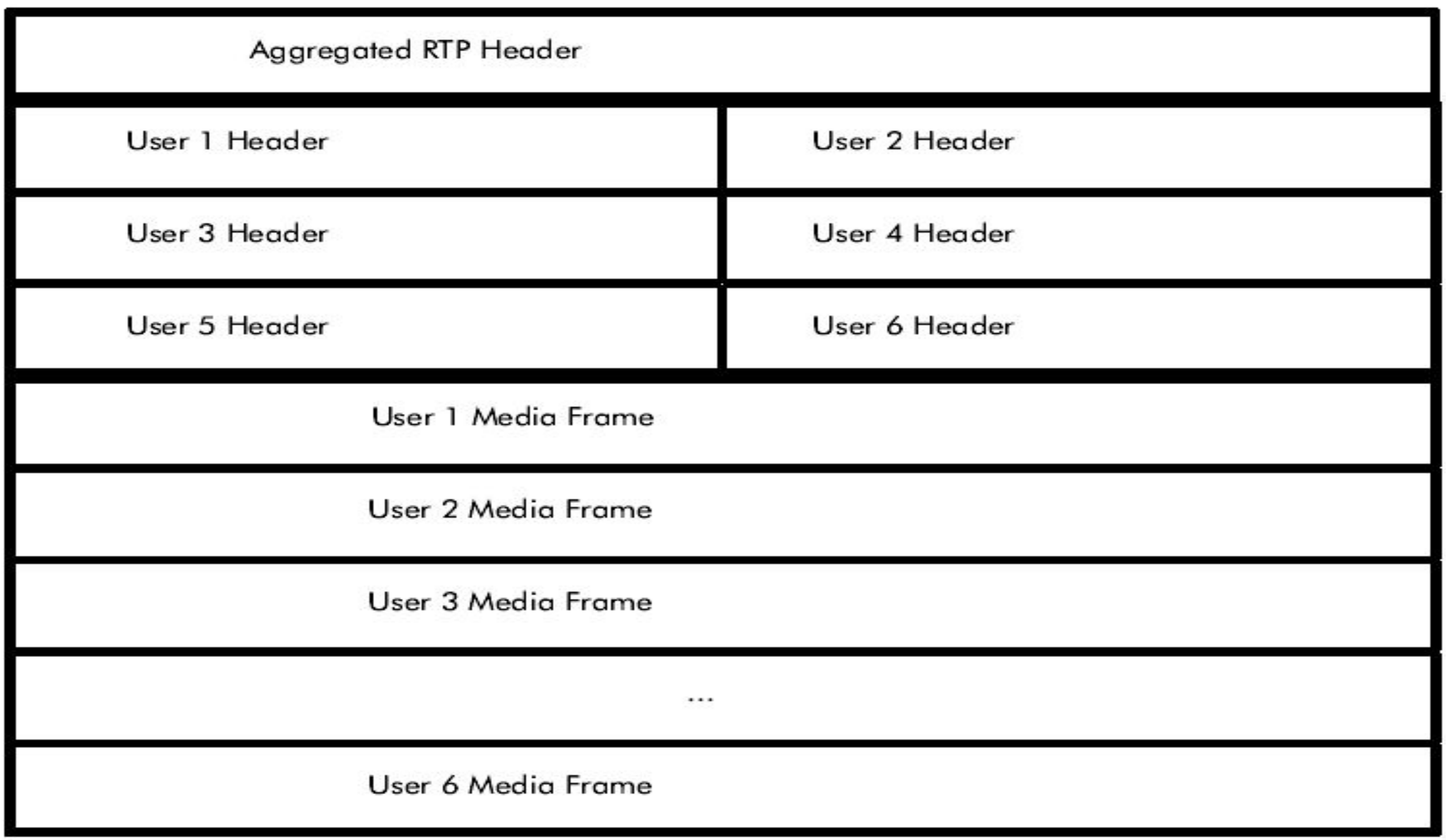
- Ethernet: 18 bytes
- IP : 20 bytes
- UDP : 8 bytes
- RTP : 12 bytes

Having an overhead of 58 bytes and a payload 72 bytes, the overhead takes 45% of the packet.

Уменьшение избыточности за счет агрегрирования RTP сессий



Структура заголовка агрегированного пакета



Функции протокола RTCP

- **Обеспечивает контроль передачи RTP – пакетов, посредством организации обратной связи между передатчиком и приемником**
- **Передает сведения о числе переданных и потерянных пакетов, значении джиттера, задержке и др.**
- **Обеспечивает передачу уточненной информации об источнике (имя, домен, E-mail, телефонный номер, месторасположение и т.д)**
- **Использует 5 типов пакетов (SR, RR, SDES, BYE,APP)**
- **Пакеты RTCP передаются значительно реже RTP пакетов (максимальная частота один пакет в 5 сек)**

Типы пакетов RTCP.

SR – Sended Report

RR – Receiver Report

SDES – Source Description

BYE – окончание участия

APP – специфические данные приложения

Report Block.

0	31
SSRC – Synchronization Source Idenifier	
Fraction lost	Packet Lost
Highest Sequence Number Received	
Interarrival Jitter	
LRS	
DLSR	

Fraction Lost (8 бит) доля потерянных пакетов данного источника относительно общего числа пакетов

Packet Lost (24 бит) – общее число потерянных пакетов данного источника

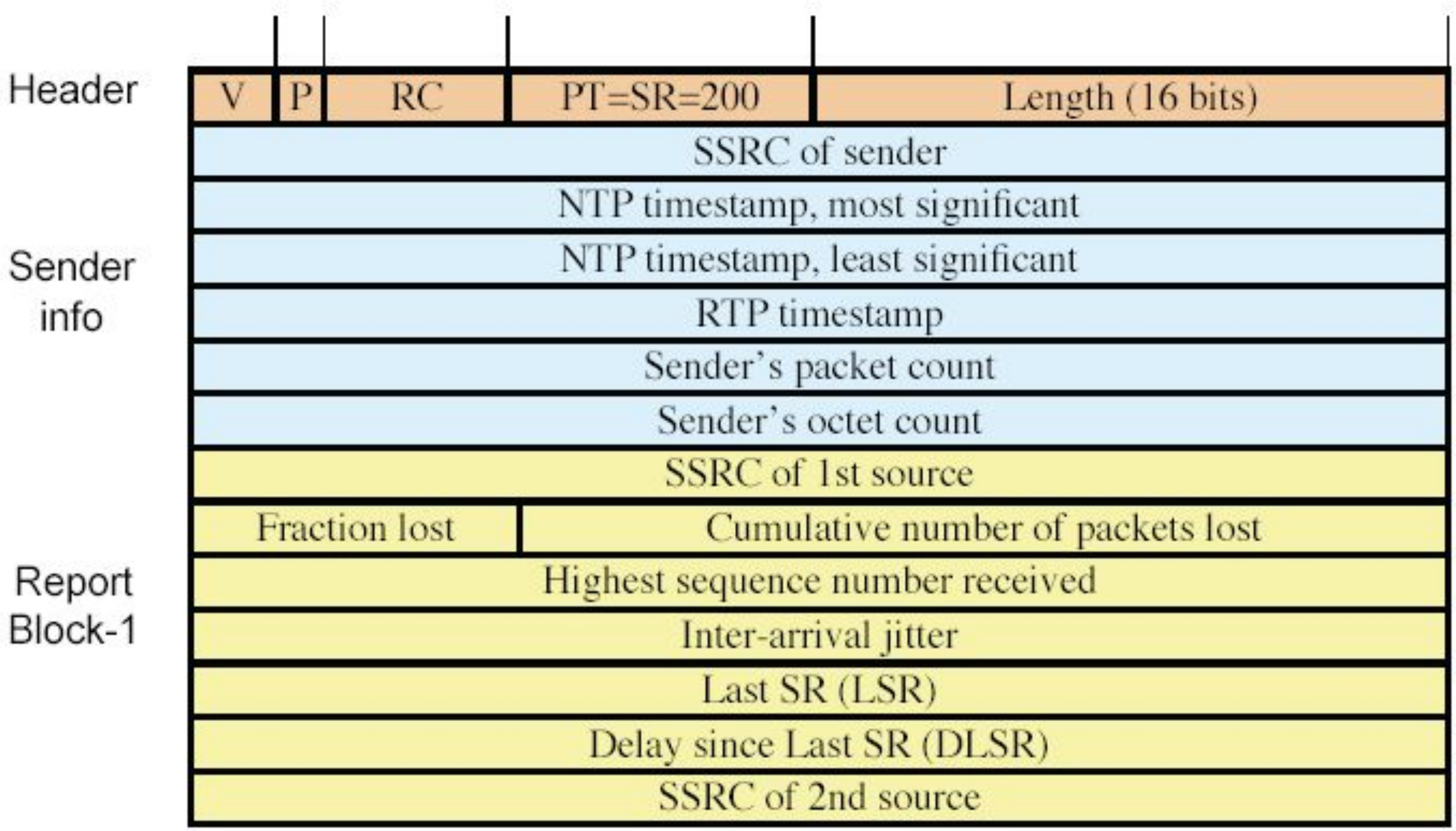
Highest Sequence Number максимальный номер пакета, полученного от данного источника

Interarrival Jitter – среднее абсолютное значение изменения времени распространения.

LSR – старшая часть последнего значения NTP TimeStamp, полученного от данного источника

DLSR – задержка времени от получения последнего сообщения от данного источника до формирования данного блока

Формат пакета Sender Report.



Назначение полей пакета SR(1)

- the 5–bit **reception reports count (RC)** , which is the number of report blocks included in this SR;
- the **packet type (PT)** is 200 for an SR;
- the 16–bit **length of this SR** including header and padding;
- the **SSRC of the originator of this SR**. This SSRC will also be in the RTP packets originated from this host.

Назначение полей пакета SR(2)

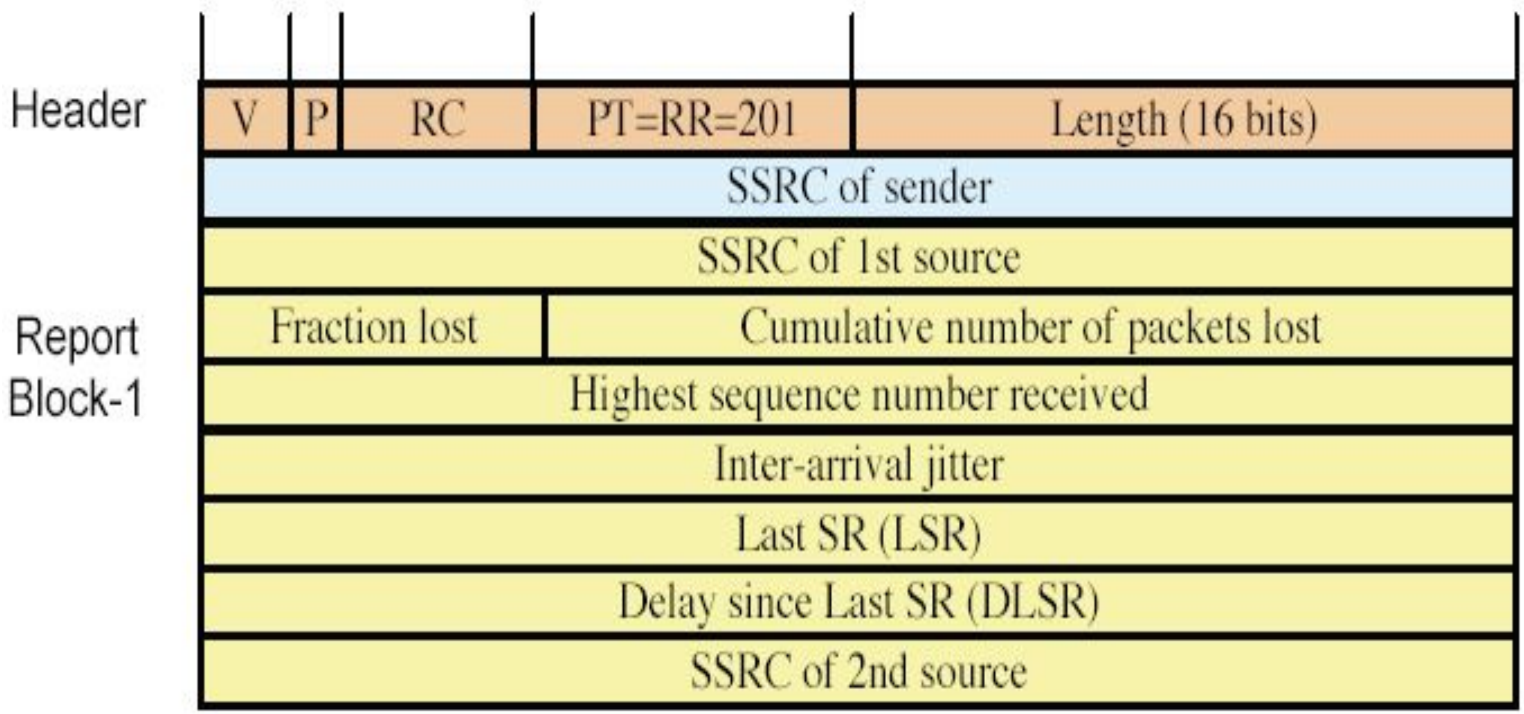
- the **NTP timestamp of the sending time** (wallclock time) of this report.

  The NTP timestamp in a sender report may be used in combination with timestamps returned in reception reports from receivers to measure round–trip propagation to those receivers. To know this absolute 'delay', the clocks of the end users must be synchronized.
- the **RTP timestamp**, which represents the same time as above, but with the same units and random offset as in the timestamps of the RTP packets (relative timing);
- **sender's packet count** (32 bits) from the beginning of this session up to this SR;
- **sender's payload octet count** (32 bits) since the beginning of the session.

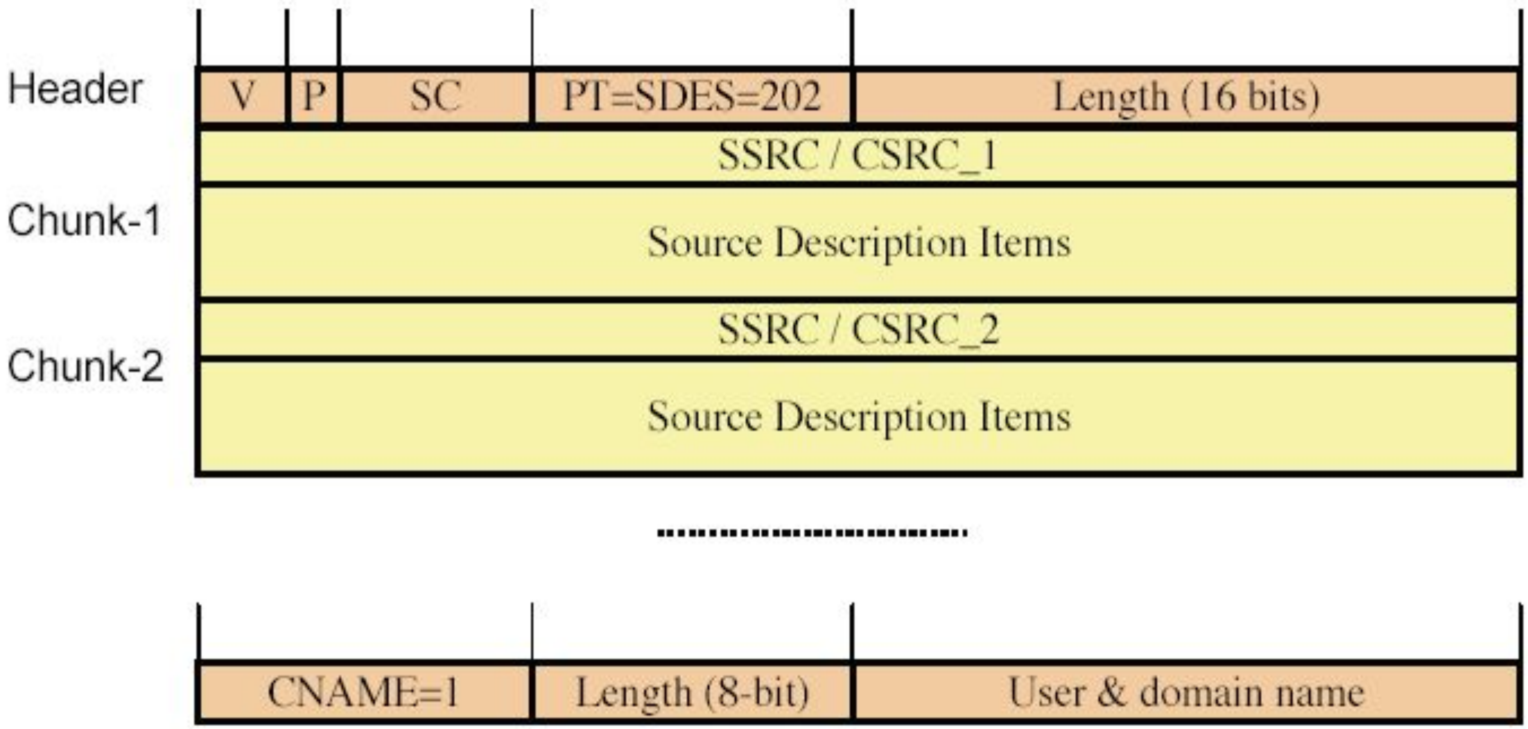
Назначение полей пакета SR(3)

- **SSRC_n** (32 bits), which is the SSRC of the source about which we are reporting;
- **fraction of packets** that were **lost** (8 bits);
- **cumulative number of packets lost** (24 bits) since the beginning of reception;
- **extended highest sequence number received** (32 bits) in an RTP data packet from this source;
- **interarrival jitter** (32 bits), which is an estimation of the variance of the interarrival time between RTP packets, measured in the same units as the RTP time stamp;
- the **last SR timestamp (LSR)** –32 bits– is represented by the middle 32 bits of the NTP timestamp of the last received SR;
- the **delay since the last SR arrived (DLSR)** –32 bits– together with the last SR timestamp, the sender of this last SR can use it to compute the round trip time.

Формат пакета Receiver Report.



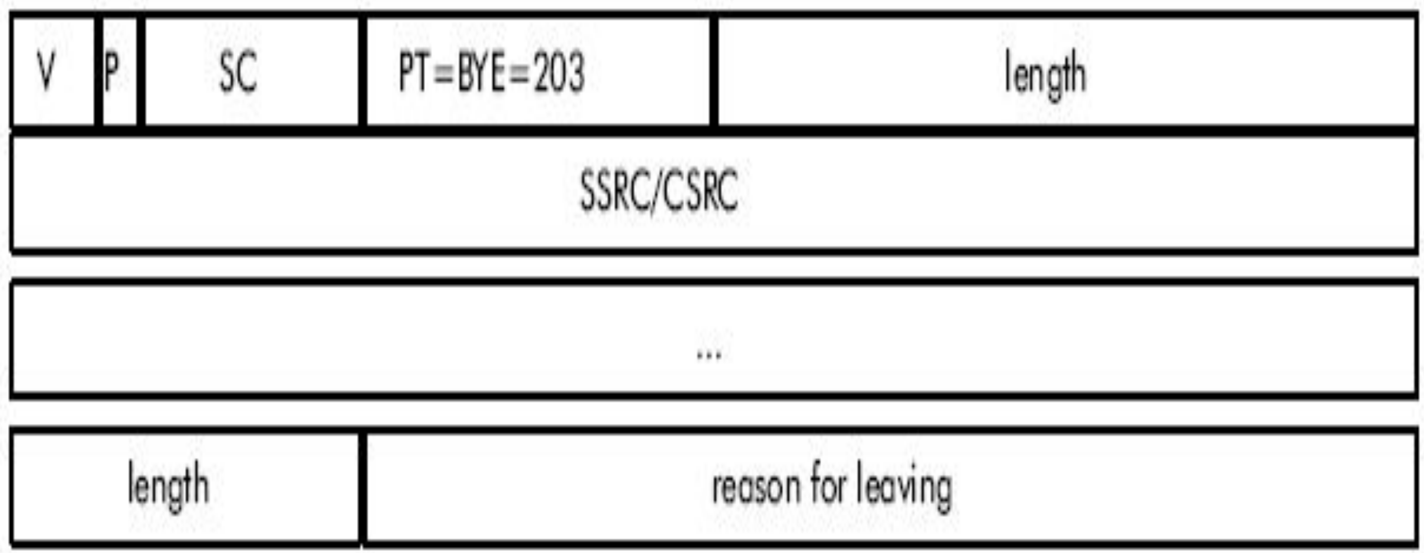
Формат пакета Source Description.



Формы описания источника (Items)

CNAME=1	length	user and domain name ...
NAME=2	length	common name of source ...
EMAIL=3	length	email address of source ...
PHONE=4	length	phone number of source ...
LOC=5	length	geographic location of site ...
TOOL=6	length	name/version of source appl . ...
NOTE=7	length	note about the source ...
PRIV=8	length	prefix string ...
...	value string ...	

Формат пакета BYE



Формат пакета APP

V	P	subtype	PT=APP=204	length
SSRC/CSRC				
name (ASCII)				
application dependent data ...				

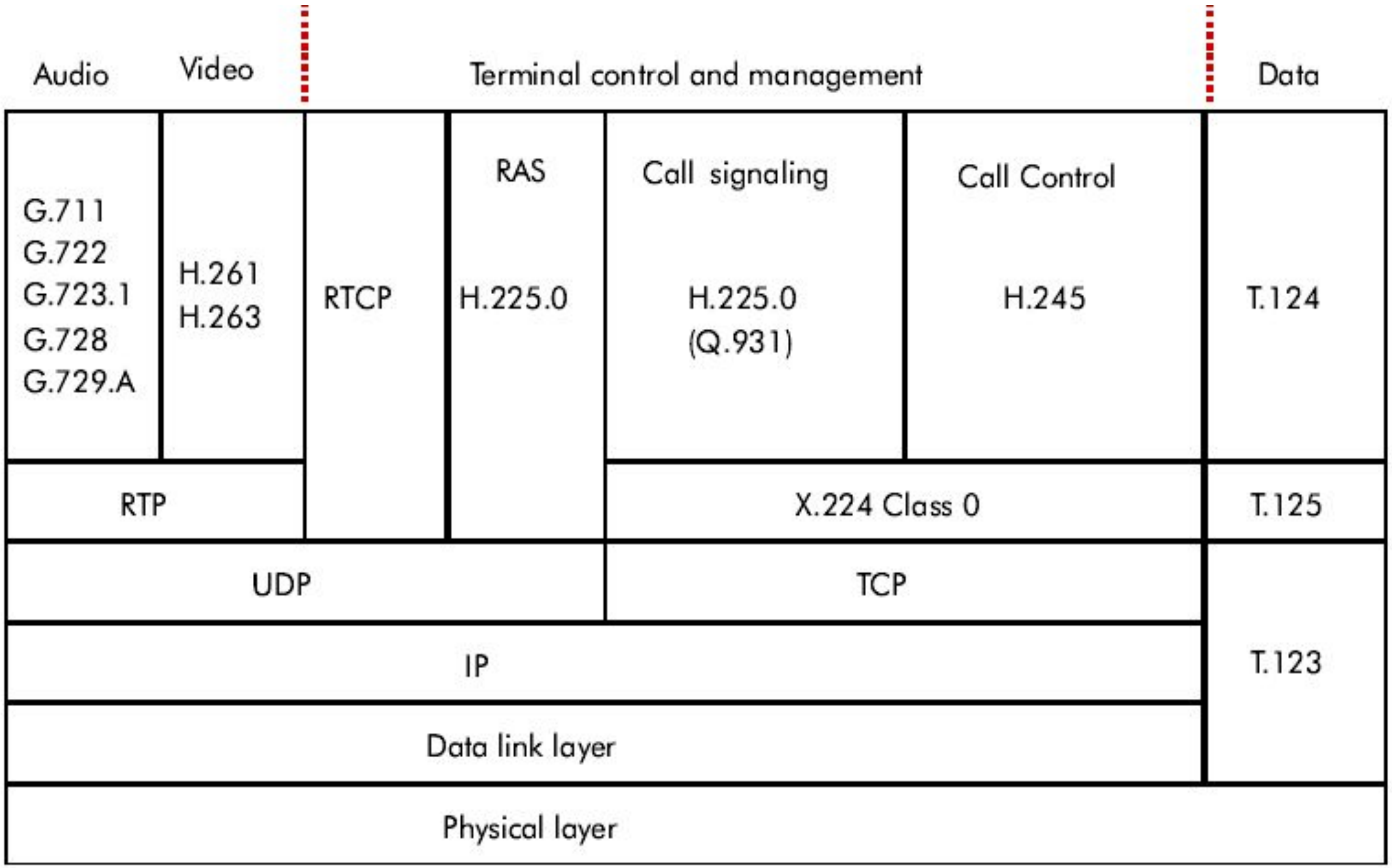
Расширение протокола RTCP XR (RFC 3611)(1)

- **Список дополнительных параметров качества VoIP:**
- **Loss Rate - доля потерянных пакетов**
- **Discard Rate - доля сброшенных пакетов из-за переполнения буфера**
- **Burst density/duration - интенсивность и длительность вспышки трафика**
- **Gap density/duration - интенсивность и продолжительность пауз (низкого уровня поступления пакетов)**
- **Round Trip Delay - задержка передачи пакета «туда и обратно»**
- **End system delay - усредненная системная задержка (с учетом асимметрии сети)**
- **Signal Level - уровень сигнала**
- **Noise Level- уровень шума**

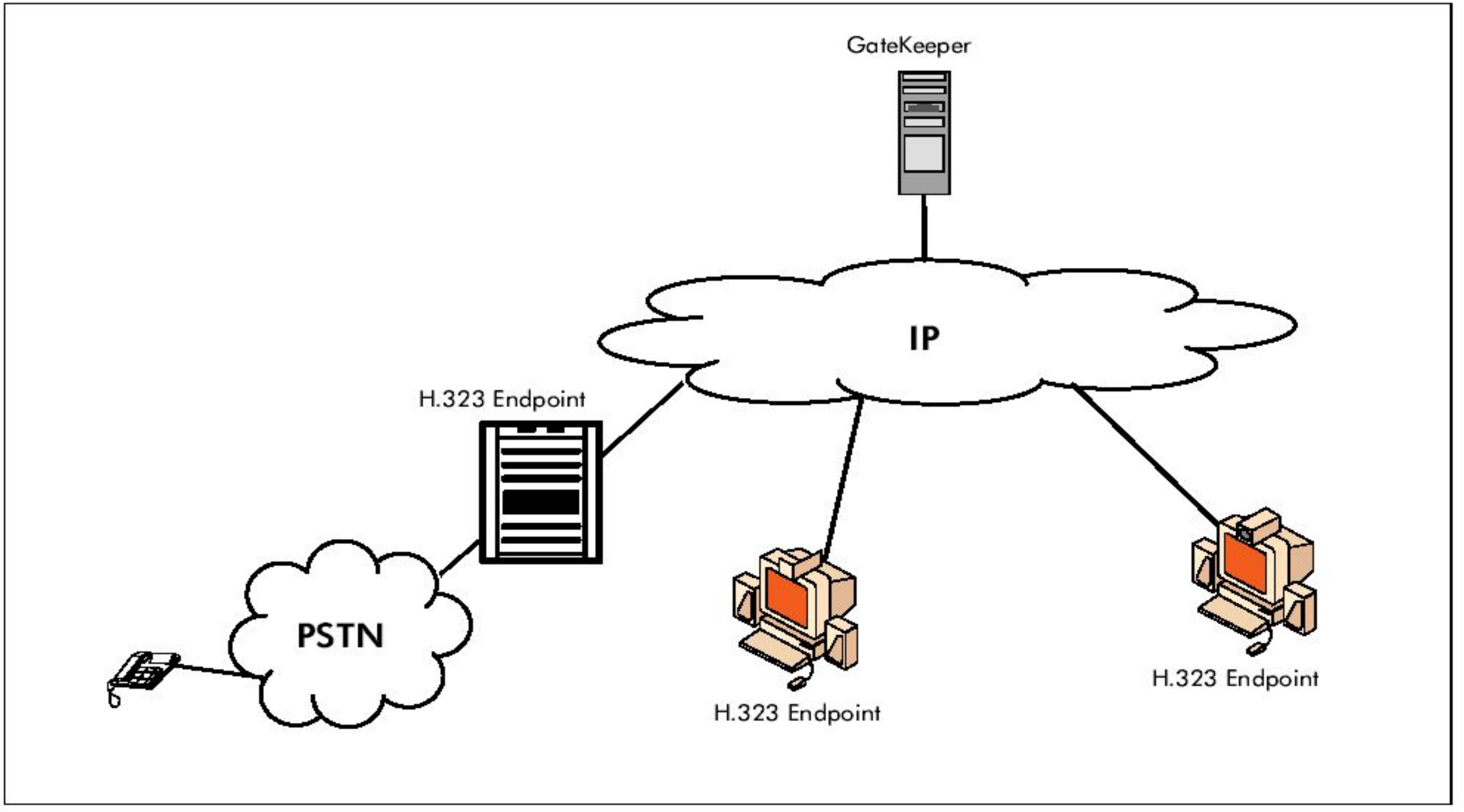
Расширение протокола RTCP XR (RFC 3611)(2)

- Residual Echo Return Loss - остаточный сигнал после работы эхо-заградителя
- R - Factor - для RTP-сеанса по Рекомендации G.107
- MOS-LQ - Estimated Mean Opinion Score for Listening Quality – экспертная оценка качества слушающим
- MOS-CQ- Estimated Mean Opinion Score for Conversational Quality - экспертная оценка качества тракта
- Gmin - допустимый порог потери пакетов
- Jitter Buffer Nominal Delay - номинальная задержка в анти-джиттерном буфере
- Jitter Buffer Maximum - максимальная задержка в анти-джиттерном буфере (зафиксированная)
- Jitter Buffer Absolute Maximum Delay – максимально допустимая задержка в анти-джиттерном буфере

Структура протоколов в H.323



Рекомендация H.323. Элементы сети.



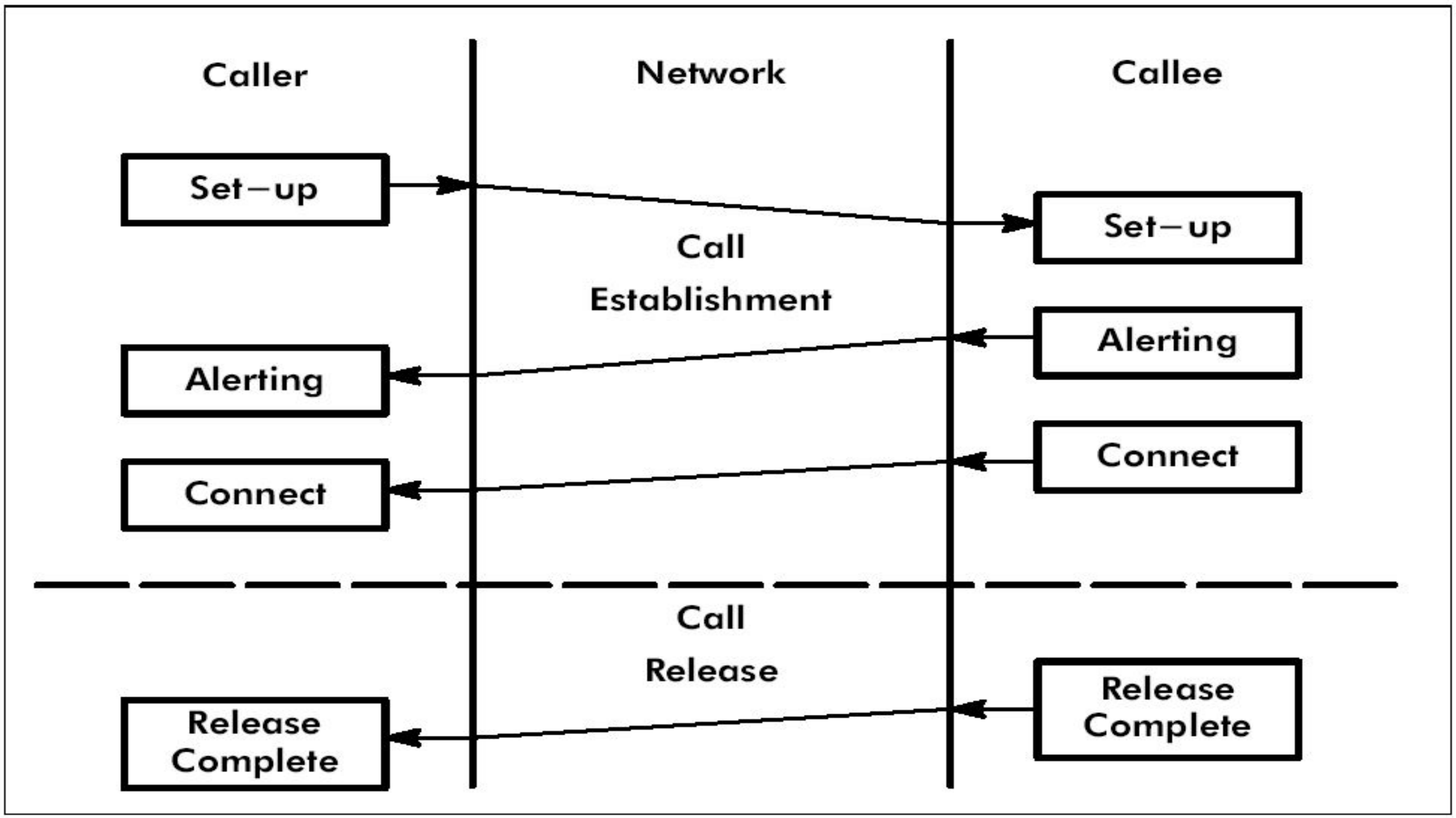
Протокол сигнализации H.225.0(Q.931)

- Q.931-like
- Used to establish a connection (Call) between two (or more) H.323 EPs
- 2 Modes:
  - Gatekeeper Routed (GRC): via GK
  - Direct Routed (DRC): directly between endpoints

Протокол регистрации,подтверждения и состояния (RAS)

- RAS signalling is used to perform
  - Registration
  - Admissions
  - Status
  - Bandwidth changes
  - disengage procedures
- Between endpoints and Gatekeeper

Процедура установления/разъединения по H.225.0(Q.931)



Протокол управления мультимедийной передачей H.245. Основные функции.

- **Capability determination**: during the capability set exchange, each terminal identifies the audio, video and data capabilities that it supports. For audio and video the capability set exchange identifies which codecs are supported.
- **Master/Slave determination**: H.245 signalling procedures are symmetric in that in that both sides have the same signalling capabilities. It is however desirable to have certain parameters specified by one side and agreed to by the other. This is accomplished by negaciating a master/slave relationship. It is the responsibility of the master to set the value for certain parameters.
- **Logical channel establishment**: logical channels are a programming convenience that allow management and control of the information exchange.

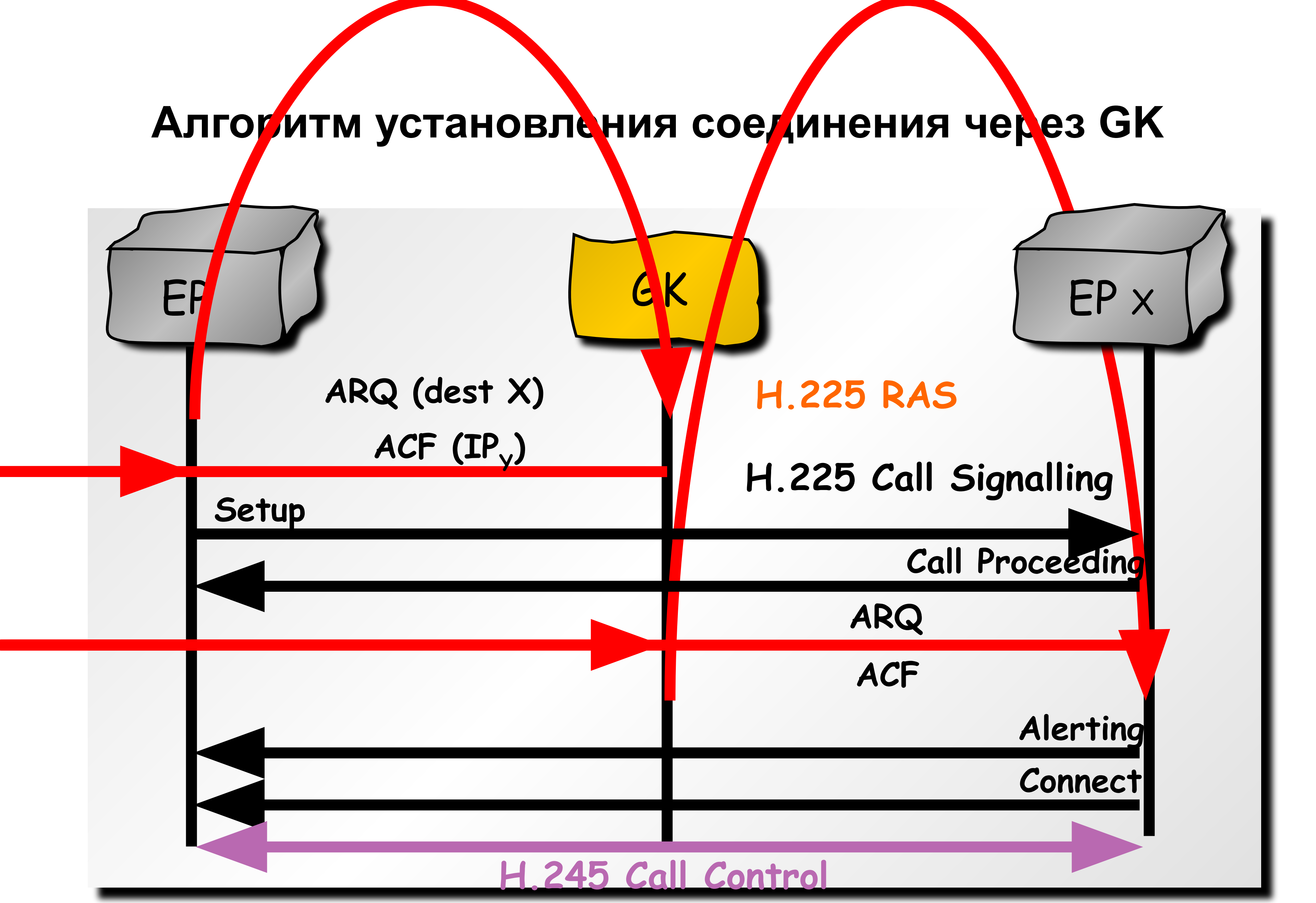
Процедура соединения. Фаза поиска и регистрации

- Before being able to make or receive calls:
  - GK Discovery (GRQ/GCF/GRJ) by EPs
    - manual
    - automatic
  - Registration with GK (RRQ/RCF/RRJ)
    - indicate to GK which IP-address is used and which aliases
      - E.164 number (telephone number)
      - H.323-ID (username@domainname)

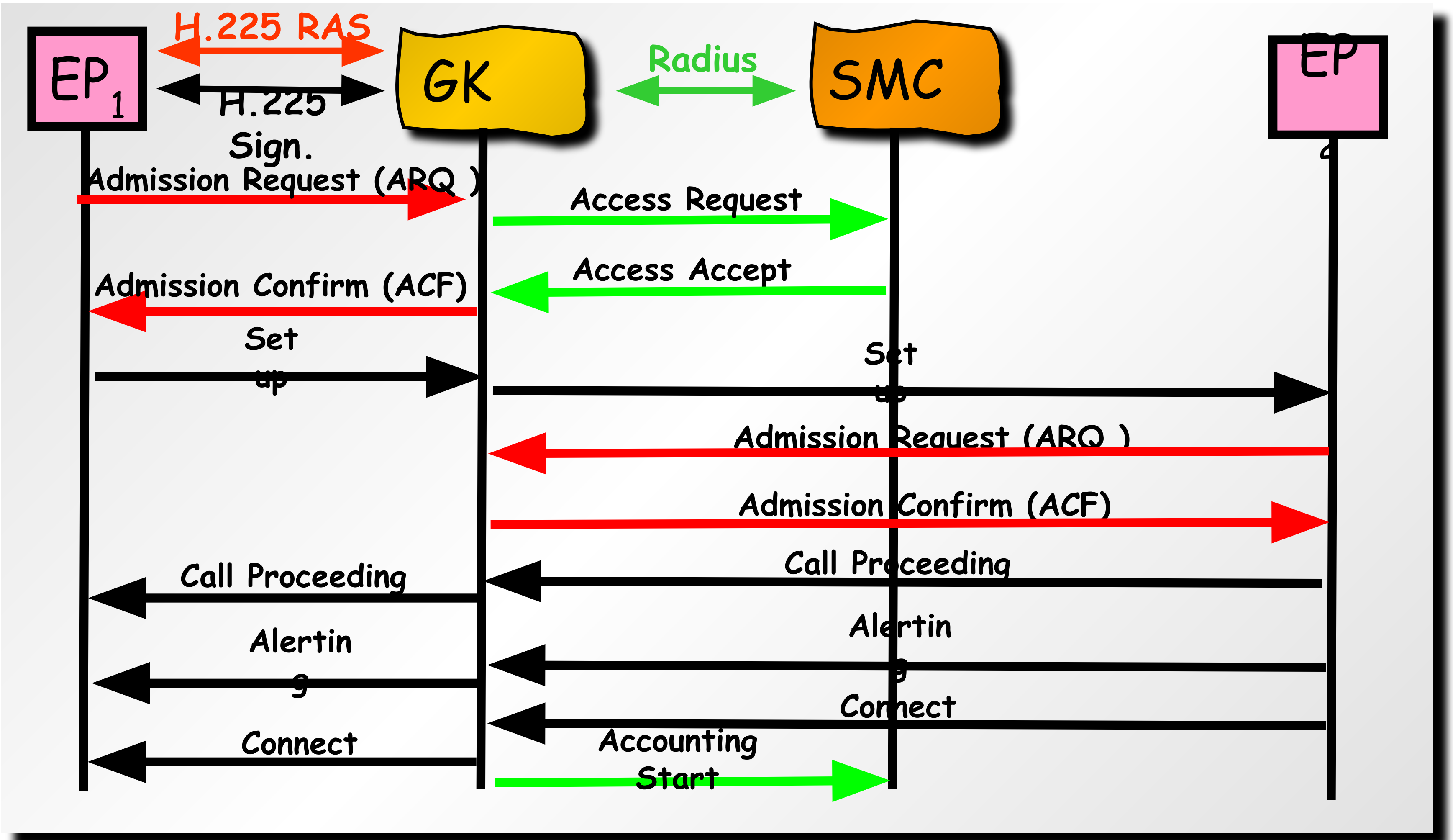
Процедура вызова. Фаза установления соединения

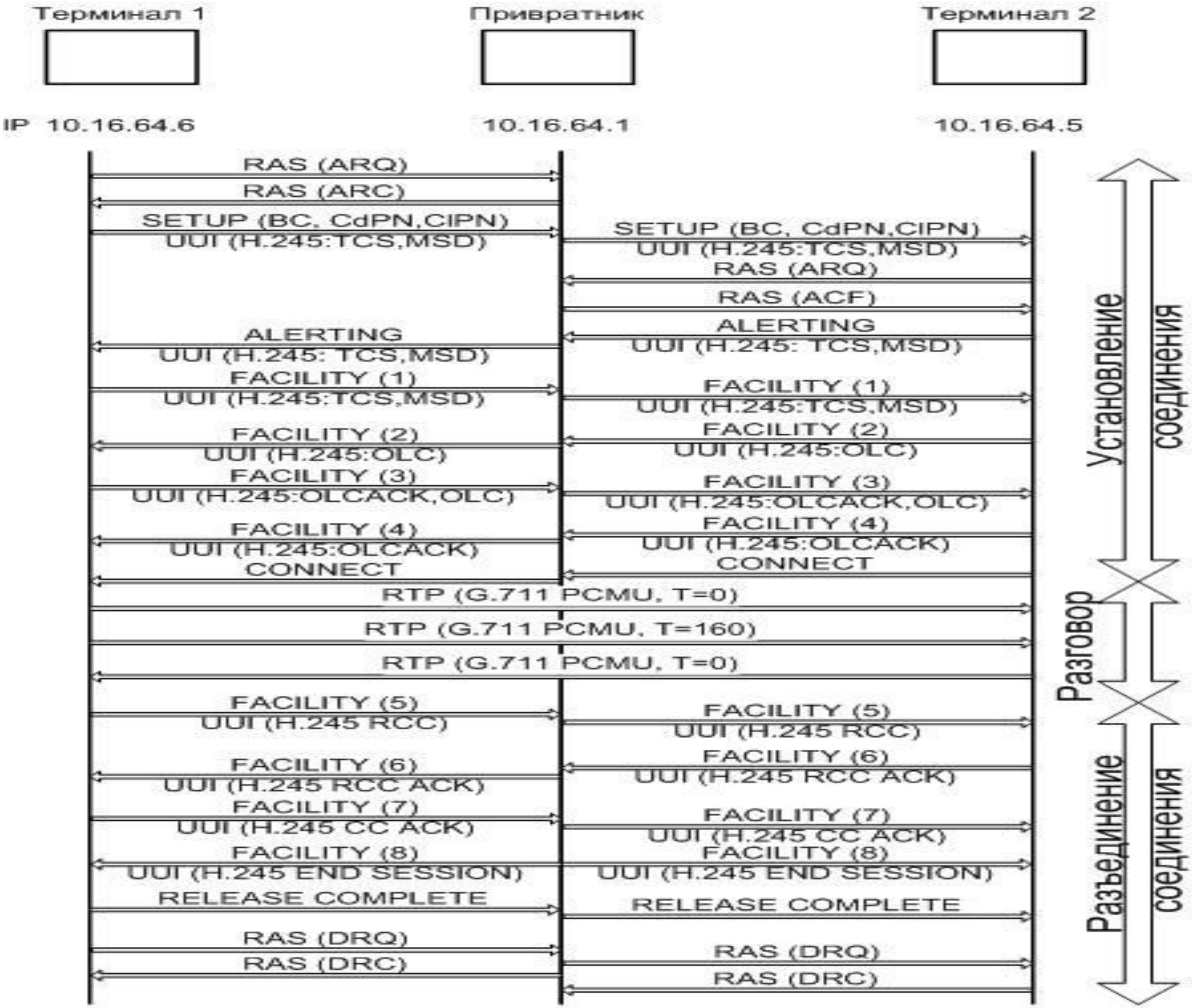
- To make calls:
  - EP asks GK permission to place a certain call (ARQ/ACF/ARJ)
    - EP specifies destination alias
    - GK replies with IP-address of destination
      - in case of DRC: IP@ is real destination
      - in case of GRC: IP@ is of GK
  - Use H.225.0 Call signalling to setup call to destination (possibly via GK)
  - Use H.245 to set up speech path

Алгоритм установления соединения через GK



Алгоритм установления соединения через SX




Терминал 1
Привратник
Терминал 2
IP 10.16.64.6
10.16.64.1
10.16.64.5
RAS (ARQ)
RAS (ARC)
SETUP (BC, CdPN,CIPN)
UUI (H.245:TCS,MSD)
SETUP (BC, CdPN,CIPN)
UUI (H.245:TCS,MSD)
RAS (ARQ)
RAS (ACF)
ALERTING
UUI (H.245: TCS,MSD)
ALERTING
UUI (H.245: TCS,MSD)
FACILITY (1)
UUI (H.245:TCS,MSD)
FACILITY (1)
UUI (H.245:TCS,MSD)
FACILITY (2)
UUI (H.245:OLC)
FACILITY (2)
UUI (H.245:OLC)
FACILITY (3)
UUI (H.245:OLCACK,OLC)
FACILITY (3)
UUI (H.245:OLCACK,OLC)
FACILITY (4)
UUI (H.245:OLCACK)
FACILITY (4)
UUI (H.245:OLCACK)
CONNECT
CONNECT
RTP (G.711 PCMU, T=0)
RTP (G.711 PCMU, T=160)
RTP (G.711 PCMU, T=0)
FACILITY (5)
UUI (H.245 RCC)
FACILITY (5)
UUI (H.245 RCC)
FACILITY (6)
UUI (H.245 RCC ACK)
FACILITY (6)
UUI (H.245 RCC ACK)
FACILITY (7)
UUI (H.245 CC ACK)
FACILITY (7)
UUI (H.245 CC ACK)
FACILITY (8)
UUI (H.245 END SESSION)
FACILITY (8)
UUI (H.245 END SESSION)
RELEASE COMPLETE
RELEASE COMPLETE
RAS (DRQ)
RAS (DRC)
RAS (DRQ)
RAS (DRC)
Установление соединения
Разговор
Разъединение соединения

Содержание сигнального сообщения SETUP(1)

- Frame 13.
- Internet Protocol, Src: 10.16.64.6 , Dst: 10.16.64.1
- Transmission Control Protocol, Src Port : 1117, Dst Port : 1721 (Seq :1, Ack :1)
- Q.931
- Protocol discriminator : Q.931
- Call reference value length : 2
- Call reference flag: Message sent from originating side
- Call reference value: 1000
- Message type: SETUP
- Bearer capability: Coding standart: ITU-T
- Information transfer capability: Unrestricted digital information
- Transfer mode: Packet mode
- User info layer 1 protocol: H.221 and H.242
- Calling party number: ‘6’
- Called party number: ‘5’

Содержание сигнального сообщения SETUP(2)

- User-User
- Protocol discriminator: X.208 and X.209 coded user information
- H.225.0 CS
- H.323_UserInformation
- H323-message-body: setup
- H245Tunneling: true
- H245Control: request: terminalCapabilitySet
- capabilityTable: 3 items
- Item 0
- receiveAudioCapability: g711Ulaw64k
- Item 1
- receiveAudioCapability: g711Alaw64k
- Item 2
- receiveAudioCapability: gsmFullRate
- H245 request: masterSlaveDetermination
- statusDeterminationNumber: 22981

Содержание сигнального сообщения ALERTING

- Frame 14.
- Internet Protocol, Src: 10.16.64.1 , Dst: 10.16.64.6
- Transmission Control Protocol, Src Port : 1721, Dst Port : 1117 (Seq :1, Ack :256)
- Q.931
- Protocol discriminator : Q.931
- Call reference value length : 2
- Call reference flag: Message sent to originating side
- Call reference value: 1000
- Message type: ALERTING
- User-User
-   Protocol discriminator: X.208 and X.209 coded user information
- H.225.0 CS
-   H.323_UserInformation
-     H323-message-body: alerting
-     H245Tunneling: true
-     H245Control: request: terminalCapabilitySet
-         capabilityTable: 3 items
-       Item 0
-       receiveAudioCapability: g711Ulaw64k
-       Item 1
-       receiveAudioCapability: g711Alaw64k
-       Item 2
-           receiveAudioCapability: gsmFullRate
-     H245 request: masterSlaveDetermination
-       statusDeterminationNumber: 234
-     H245 response: masterslaveDeterminationAck
-       Decision: slave

Содержание сигнального сообщения FACILITY(1)

- Frame 15.
- Internet Protocol, Src: 10.16.64.6 , Dst: 10.16.64.1
- Transmission Control Protocol, Src Port : 1117, Dst Port : 1721 (Seq :256, Ack :208)
- Q.931
- Protocol discriminator : Q.931
- Call reference value length : 2
- Call reference flag: Message sent from originating side
- Call reference value: 1000
- Message type: FACILITY
- User-User
- Protocol discriminator: X.208 and X.209 coded user information
- H.225.0 CS
- H.323_UserInformation
- H323-message-body: facility
- H245Tunneling: true
- H245Control: responce: terminalCapabilitySetAck
- H245 response: masterslaveDeterminationAck
- Decision: master

Недостатки H.323

- Сложность протокольного стека
- Необходимость поддержки двух стеков TCP/IP и UDP/IP
- Плохая масштабируемость
- Трудности при перенаправлении вызовов