# Arquitetura de Softswitch

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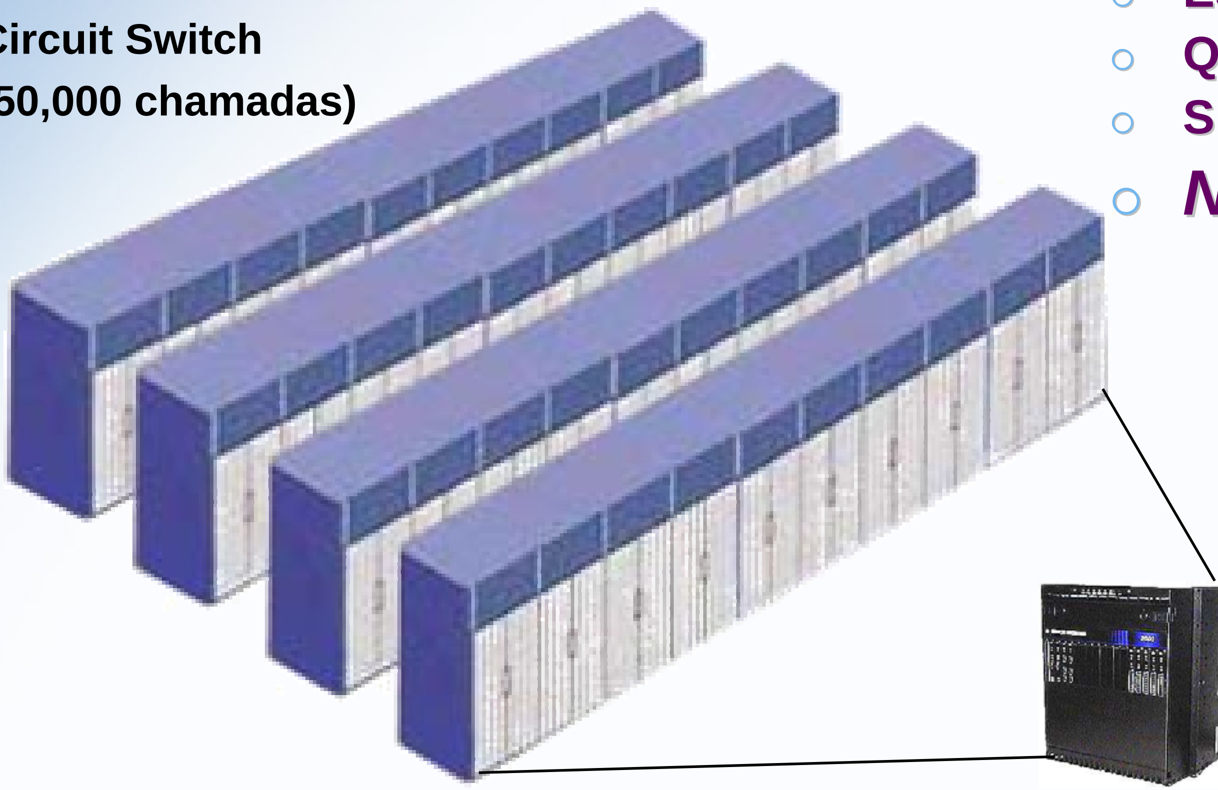
# O poder de um “Softswitch”

- Confiabilidade
- Escalabilidade
- Qualidade de Serviço
- Sinalização
- ***Novas Aplicações***

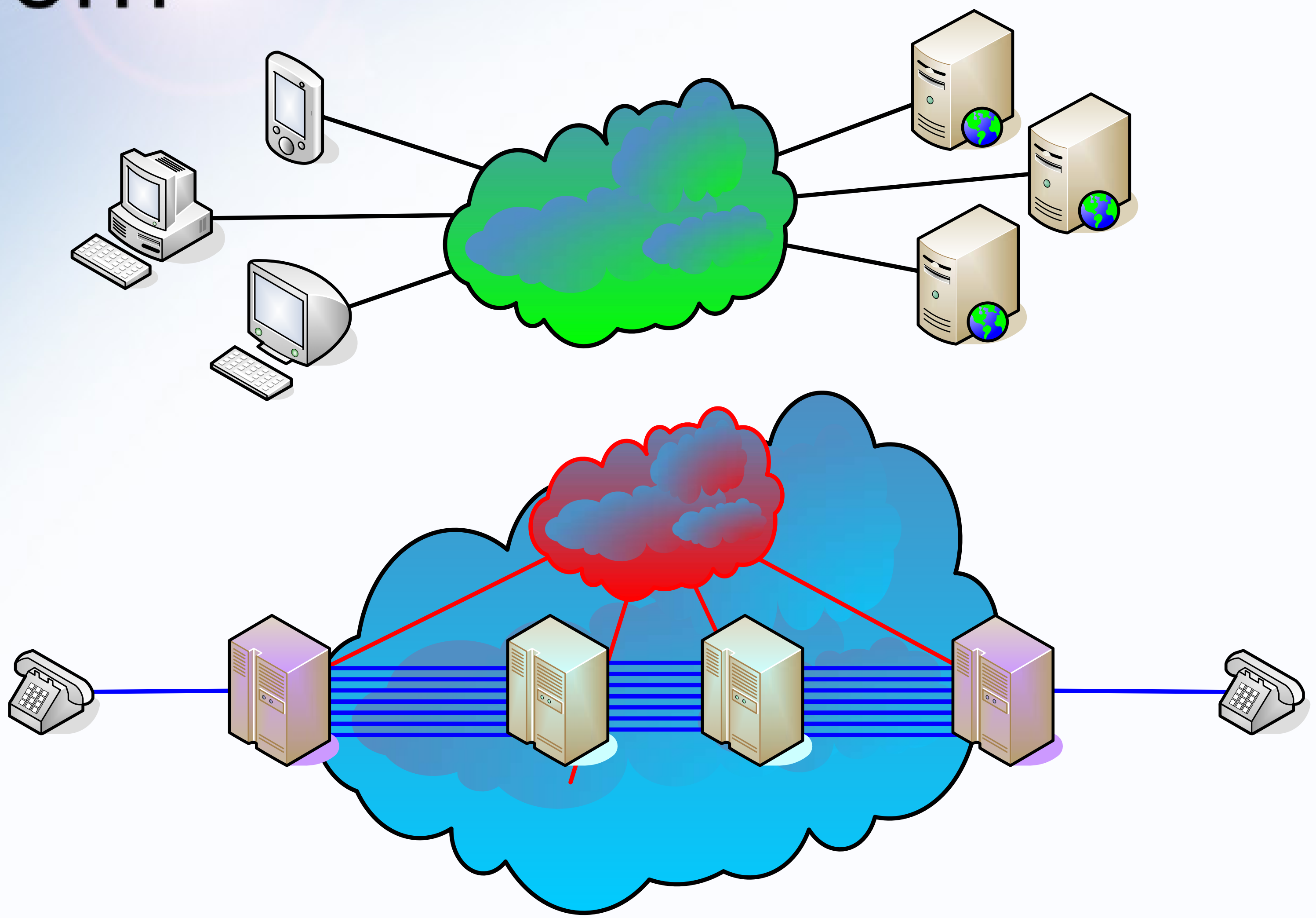
**Circuit Switch**
**(50,000 chamadas)**



**Softswitch**
**(50,000+ chamadas)**

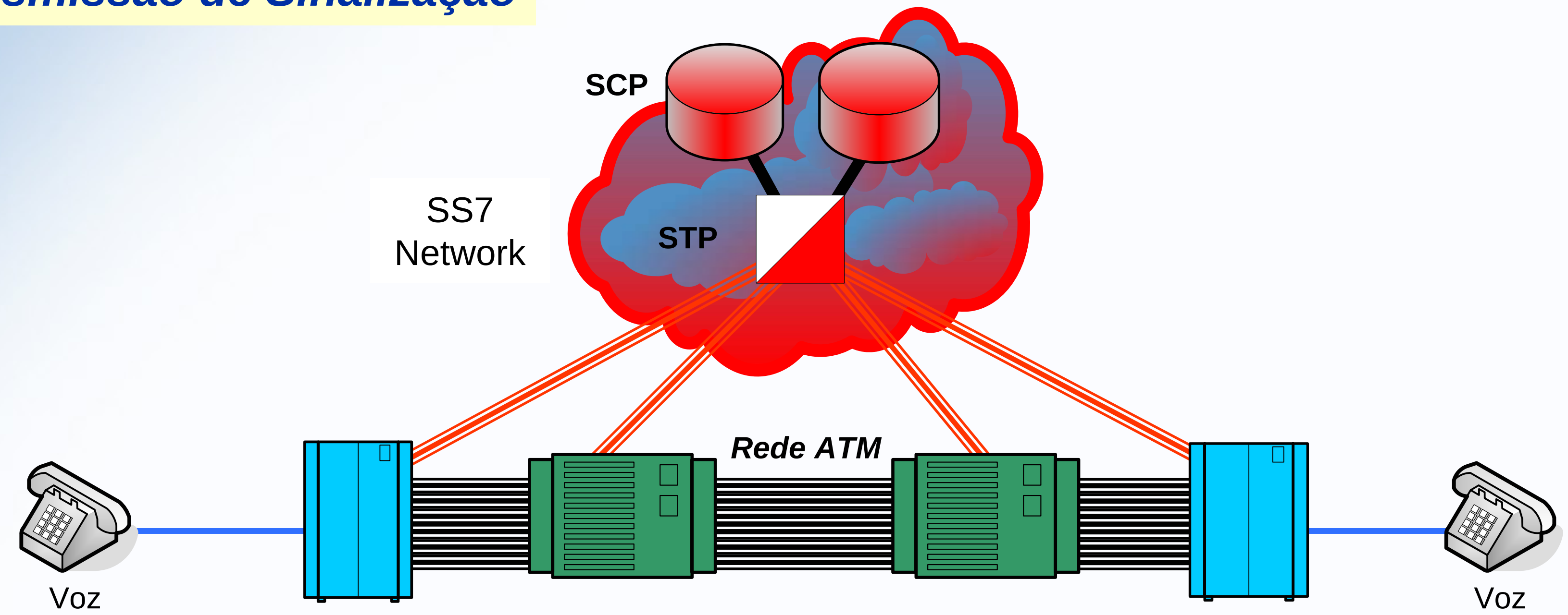
# *A Rede Pública de Telefonia*

# Relacionamento Tradicional Redes de Voz e Dados



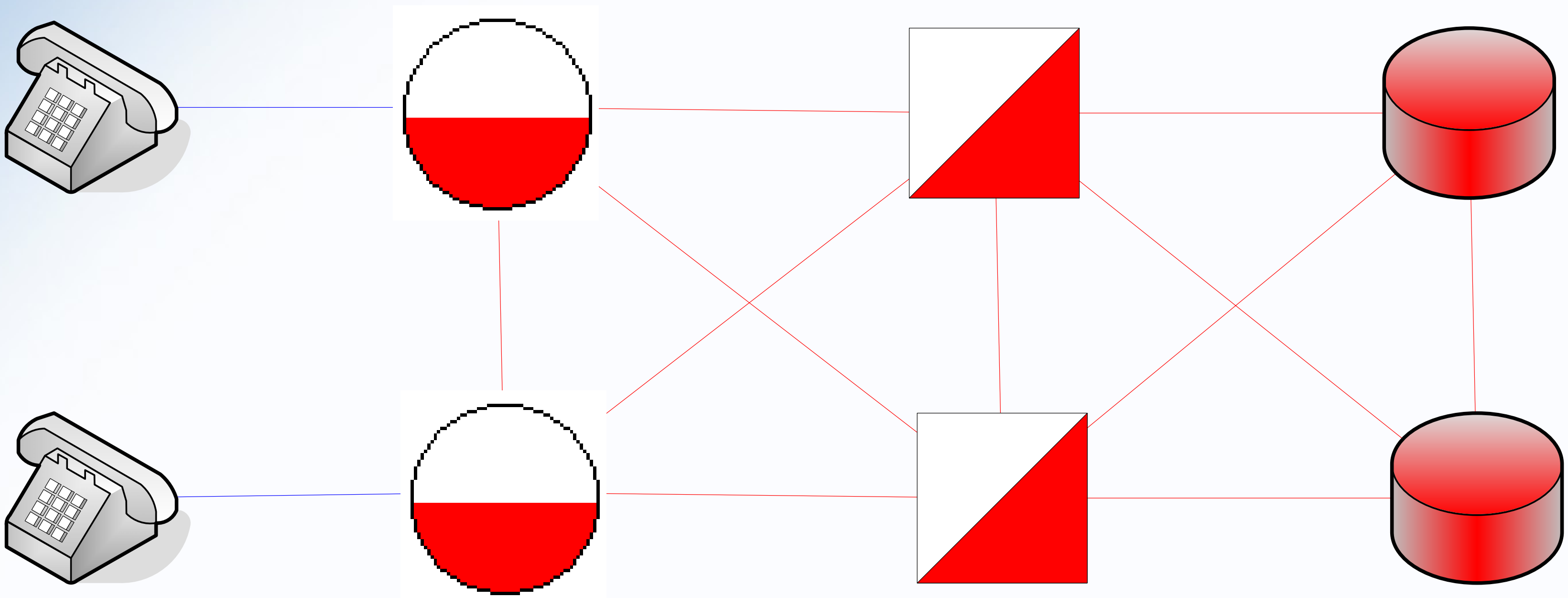
# Relação entres Switches (C4&C5)

***Transmissão da Voz***
***Transmissão de Sinalização***

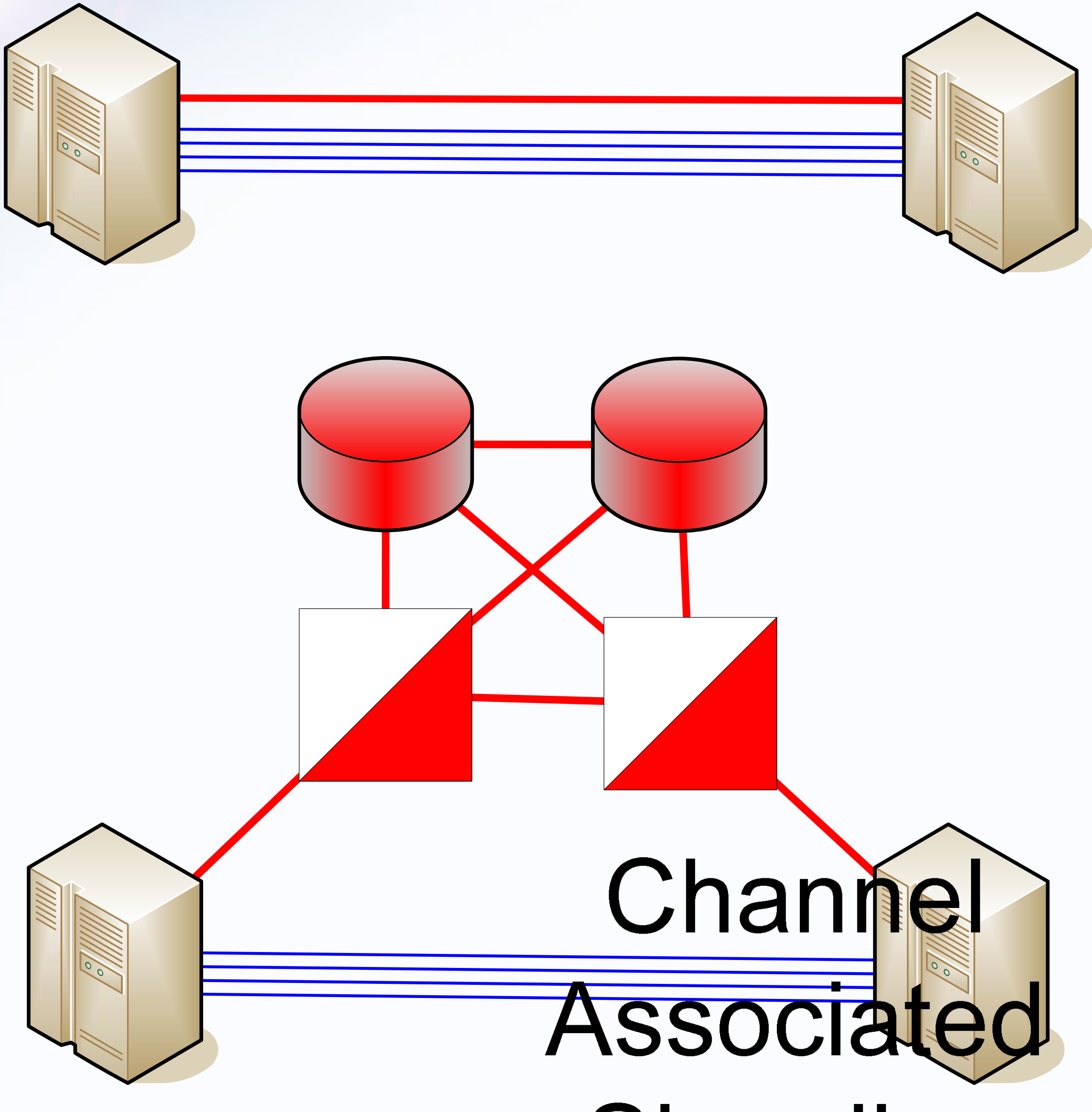



**Cada Canal de Voz = 4000Hz de Banda**
**4000Hz Digitalizados = 1 PCM = 64kbps = G.711**

# Pontos de Sinalização – SS7



# Redes CAS & CCS



# Descrição dos Codecs de Voz Padrões ITU

WAVEFORM CODECS (menor delay, maior banda)

- **G.711** PCM das frequências de voz
- G.726 ADPCM (Adaptative Differential PCM)

Analysis-by-Synthesis (Abs) CODECS (maior delay, menor banda)

- G.728 Low-Delay CELP(LD-CELP)
- **G.723.1** Algebraic Code-Excited Linear Prediction (ACELP)
- **G.729** Coder de voz a 8Kbps
  - *G.729 Annex A (Reduz complexidade)*
  - *G.729 Annex B*
  - *G.729 Annex D*
  - *G.729 Annex E*

# Descrição dos Codecs de Voz Padrões ITU

WAVEFORM CODECS

| *Padrão* | *Banda* | *Delay* | *MOS* |
|---|---|---|---|
| **G.711** | 64Kbps | 0.125ms | 4.8 |
| G.726 | 16,24,32,40Kbps | 0.125ms | 4.2 |

Analysis-by-Synthesis (Abs) CODECS

| *Padrão* | *Banda* | *Delay* | *MOS* |
|---|---|---|---|
| G.728 | 16Kbps | 2.5ms | 4.2 |
| **G.723.1** | 8Kbps | 10ms | 4.2 |
| **G.729** | 5.3,6.3Kbps | 30ms | 3.5 |

# Descrição dos Meios e Transmissão

**Rede de Transporte -> Predominante em tecnologia ATM**
**1x DS0 = 1 PCM = 64Kbps = 1 Canal de Voz**

| *OC level* | *Signal level* | *Sonet/STS* | *SDH/STM* | *DS0´s* |
| --- | --- | --- | --- | --- |
| OC-1 | 51.84Mbps | STS-1 | | 672 |
| OC-3 | 155.52Mbps | STS-3 | STM-1 | 2016 |
| OC-12 | 622.08Mbps | STS-12 | STM-4 | 8063 |
| OC-48 | 2.48Gbps | STS-48 | STM-16 | 32256 |
| OC-192 | 9.95Gbps | STS-192 | STM-64 | 129024 |
| OC-768 | 52.87Gbps | STS-768 | | 516096 |

# Protocolos de Voice Over IP

# O que é VOIP ?

**VOIP = Voice Over IP.**

**“ Capacidade de transmitir VOZ através de um meio de transmissão do tipo rede de dados, usando os protocolos do padrão TCP/IP “**

**Porque Usar VOIP ?????**

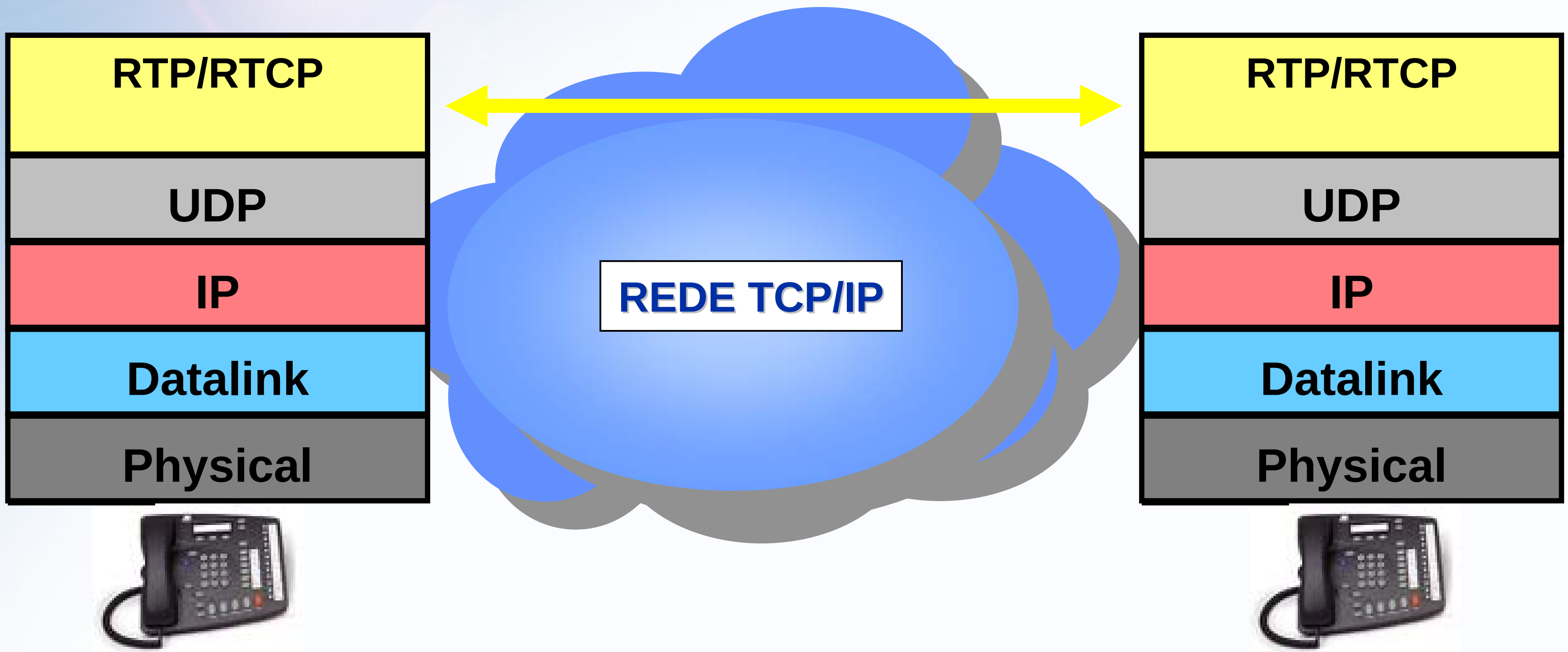
- **Redução da Banda de Transmissão (compressão da voz).**
- **Melhor utilização da infra-estrutura de transmissão (Voz, Dados e Vídeo).**
- **Menor custo de Equipamentos  e Infra-estrutura.**
- **Integração de Aplicações de Dados e Voz.**
- **Configuração Simplificada da Rede de Transmissão.**
- **Menores custos de Transmissão (DWDM)**

# Modelo Telefonia em Rede

**Componentes críticos do sitema de Telefonia em Rede:**

1) ***Protocolos de Transmissão da VOZ.***
2) ***Protocolos de Sinalização.***

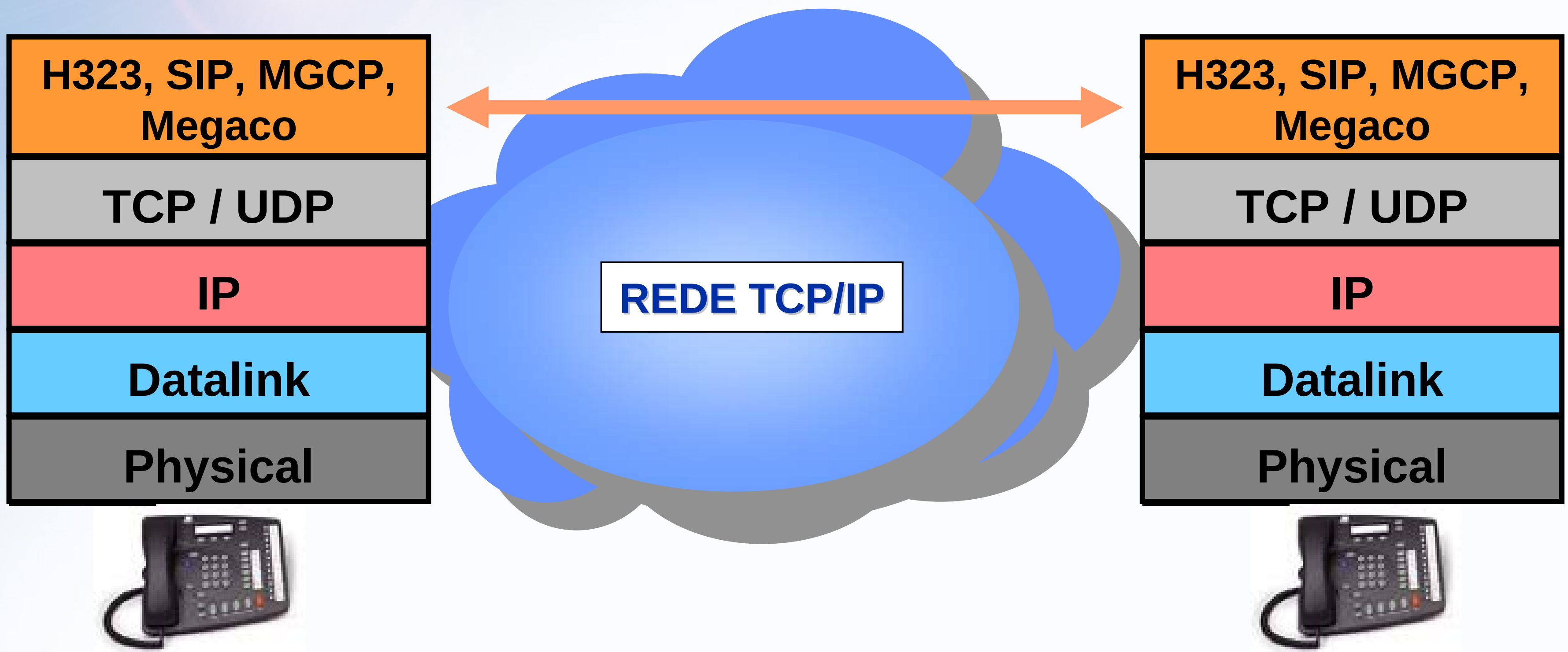
# Modelo VOIP
# Protocolos de Transmissão




**RTP/RTCP: Protocolos de Transmissão em Tempo Real de Dados .**
**Trabalham sobre UDP (Não Orientado a Conexão).**

**CODEC´s:**
**G711: 64Kbps / Modem, VOZ|, FAX**
**G729A: 9Kbps / Voz, Fax**
**G723.1: 5.3Kbps / Voz, Fax**

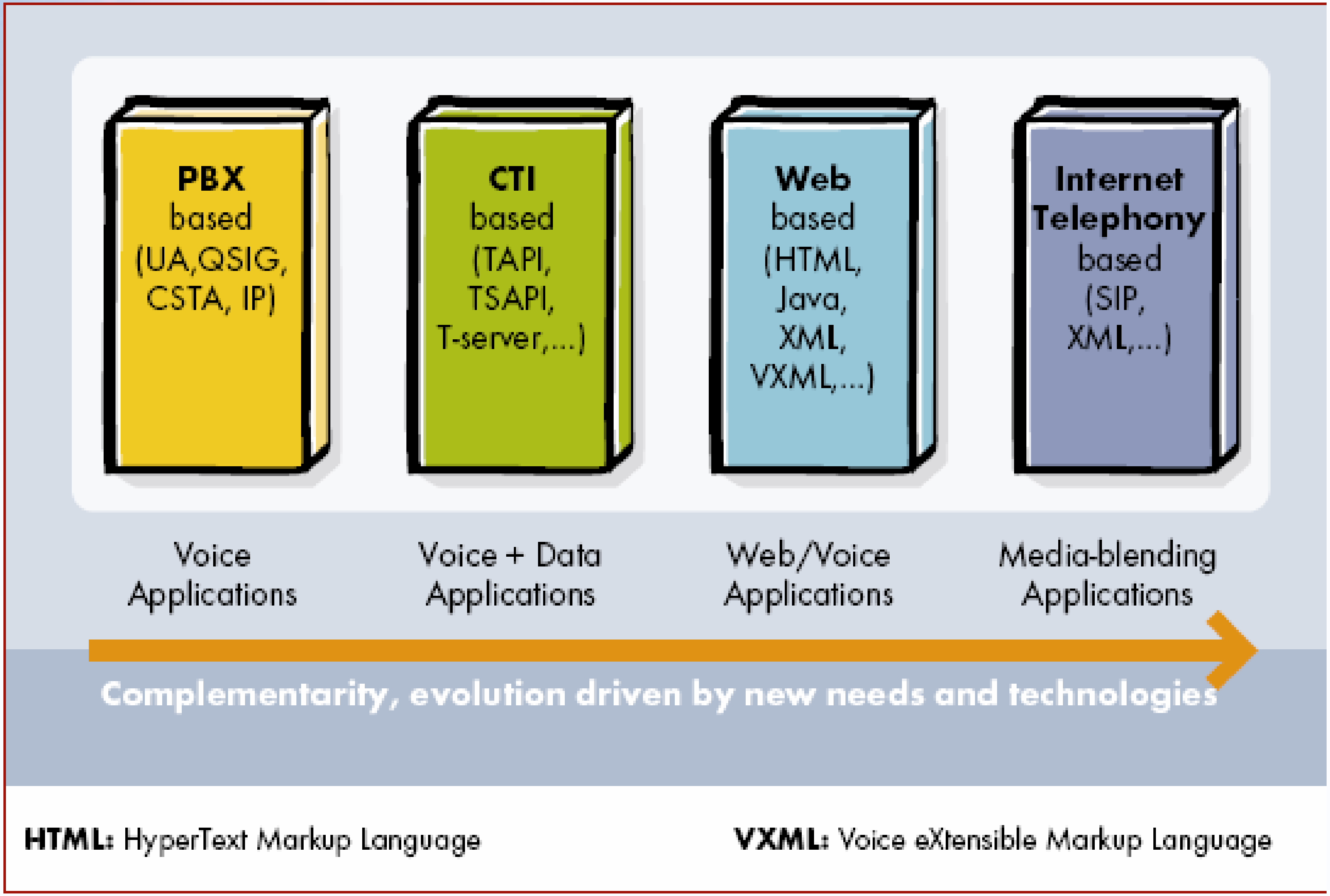
# Modelo VOIP
# Protocolos de Sinalização




**Protocolos Ponto-a-Ponto:**
- **H.323.**
- **SIP.**

**Protocolos de Controle de Gateways (Mestre/Escravo):**
- **MGCP.**
- **H.248(MEGACO).**

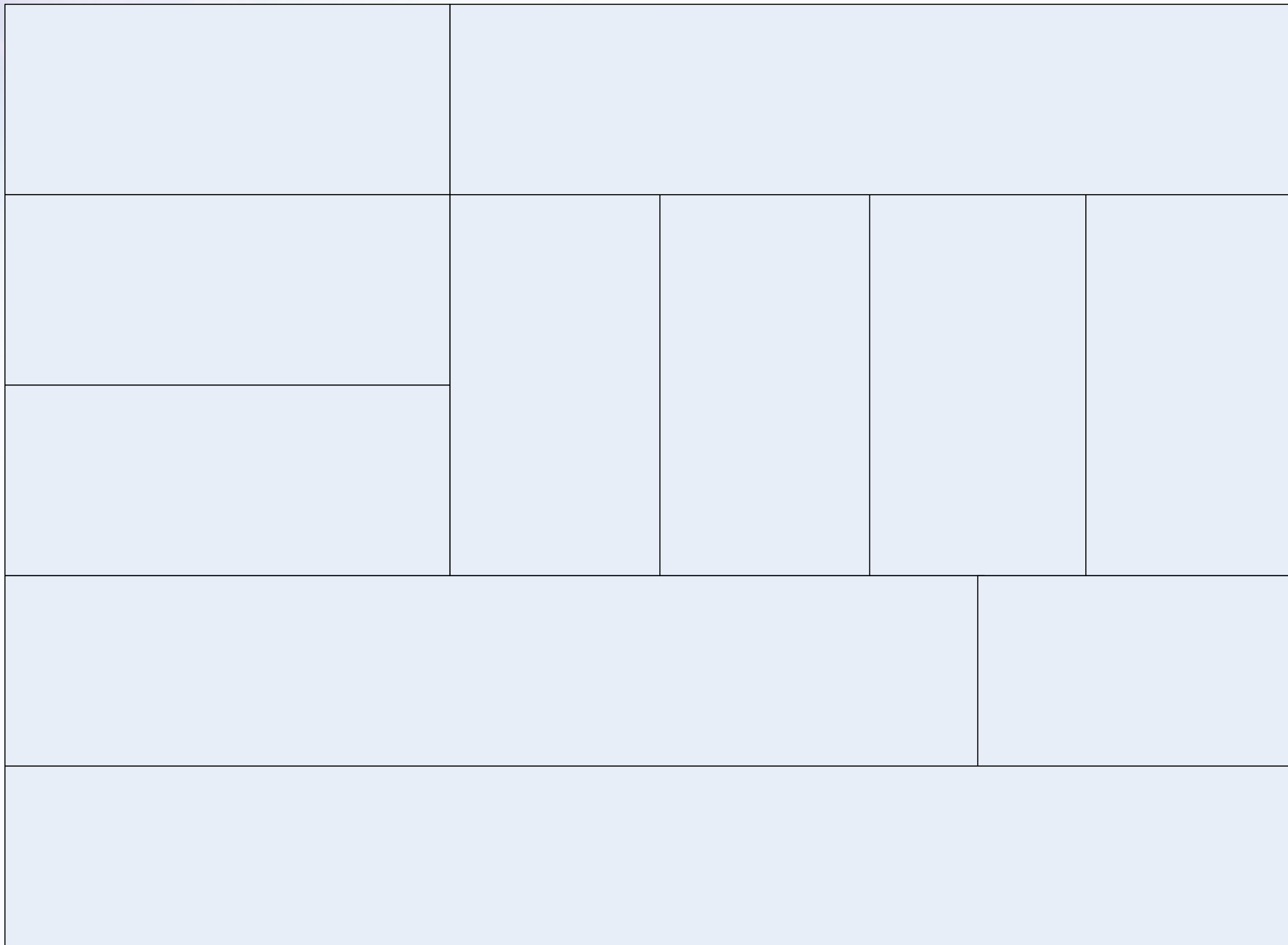
# Evolução das Aplicações de Comunicação



# Padrão H.323

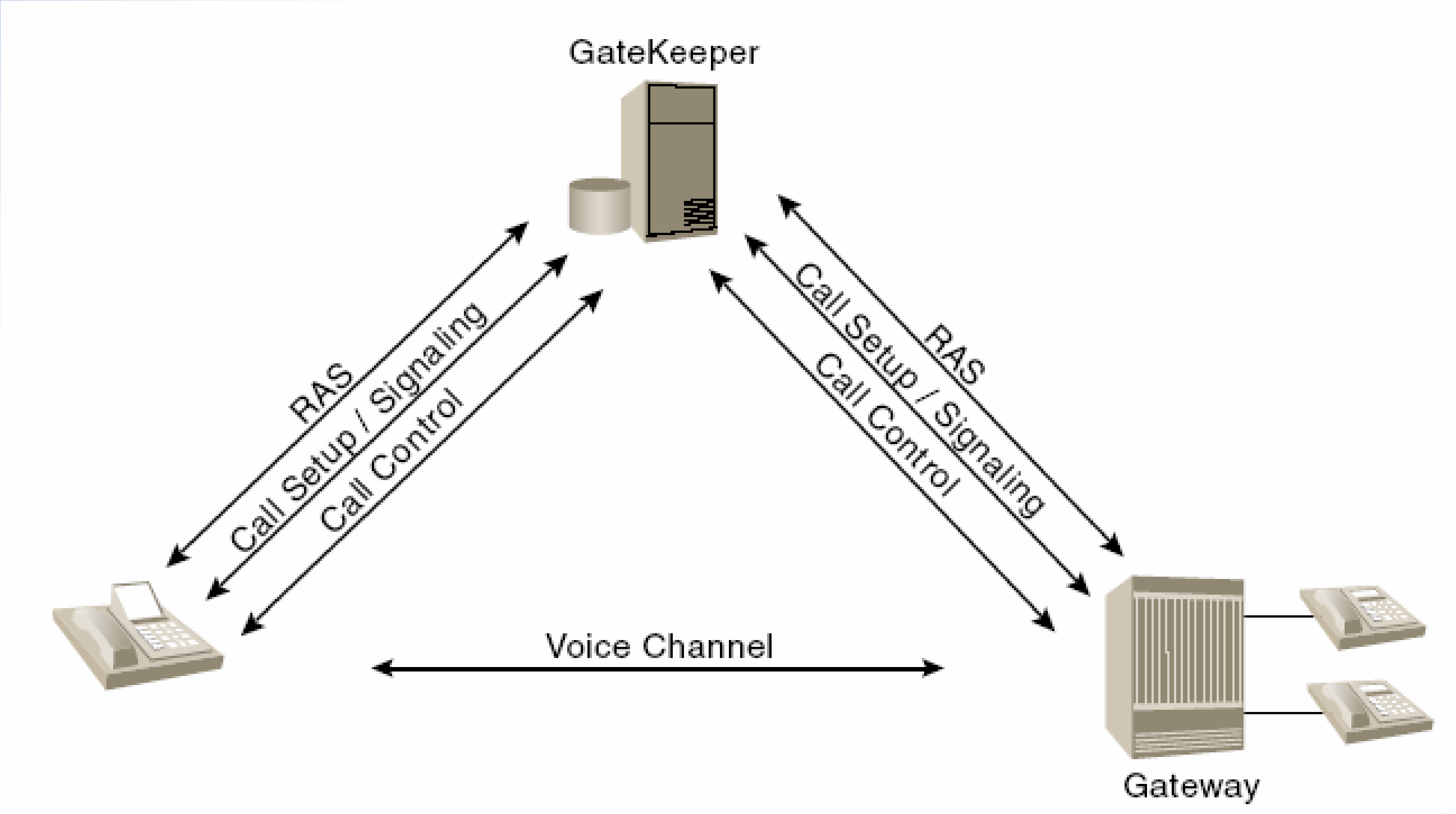
- Audio codecs (G.711, G.723.1, G.729, etc.) and video codecs (H.261, H.263) compress and decompress media streams
- Media streams transported on RTP/RTCP
  - RTP carries actual media
  - RTCP carries status and control information
- RTP/RTCP carried unreliably on UDP
- Signalling is transported reliably over TCP
  - RAS - registration, admission, status
  - Q.931 - call setup and termination
  - H.245 - capabilities exchange

# A Arquitetura do H.323



Ap

# Gatekeeper Routed Call Signalling (Q.931/H.245)



# Session Initiation protocol (SIP)

- Audio codecs (G.711, G.723.1, G.729, etc.) compress and decompress media streams
- Media streams transported on RTP/RTCP
  - RTP carries actual media
  - RTCP carries status and control information
- RTP/RTCP carried unreliably on UDP
- Signalling is transported over UDP
  - SIP – messaging mechanisms for establishment of sessions
  - SAP – Session Announcement Protocol.
  - SDP – structured language for describing the session (Mbone based)

# A Arquitetura do SIP

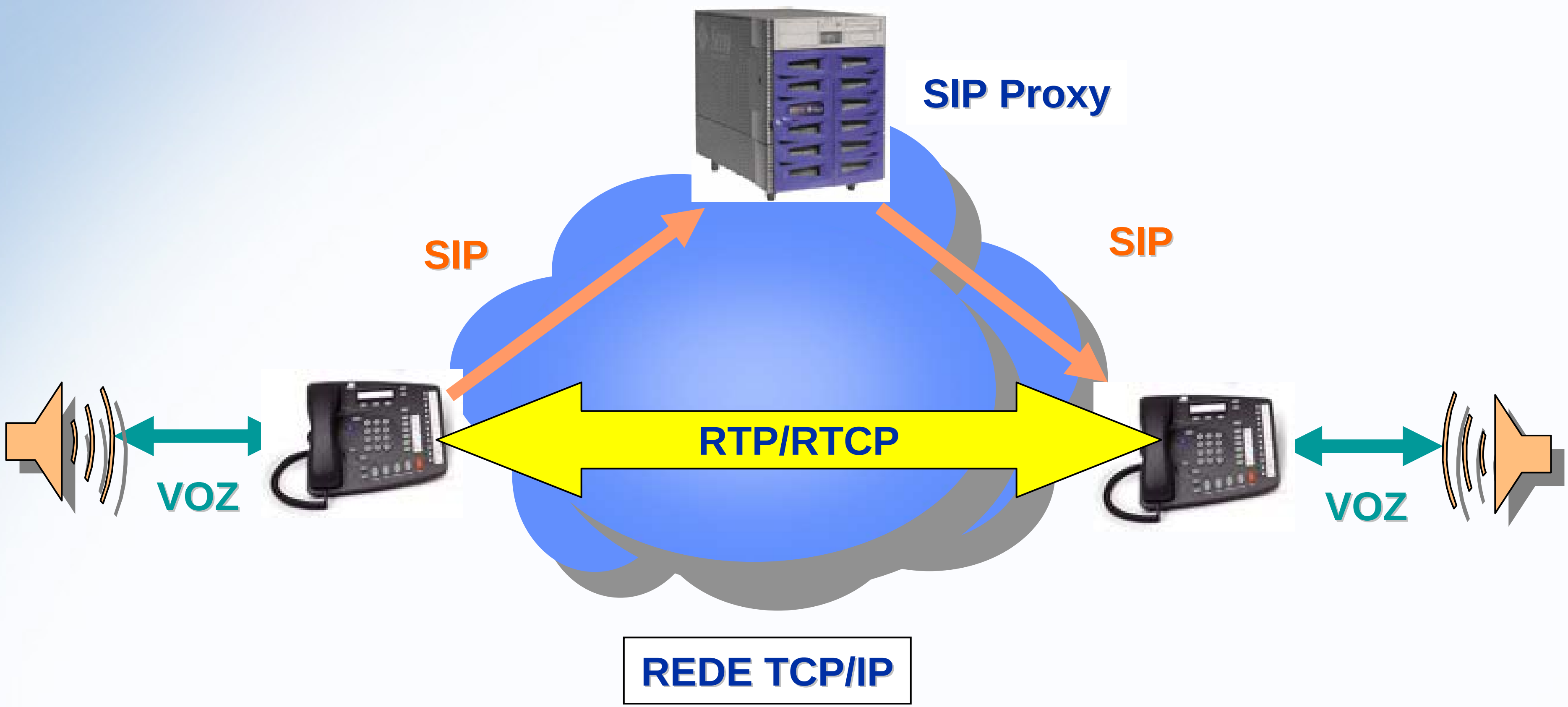
# Elementos do SIP

- **User Agent**
  - UA Client (cria as chamadas)
  - UA Server (aguarda novas chamadas).
  - Implementações de Hardware e Software

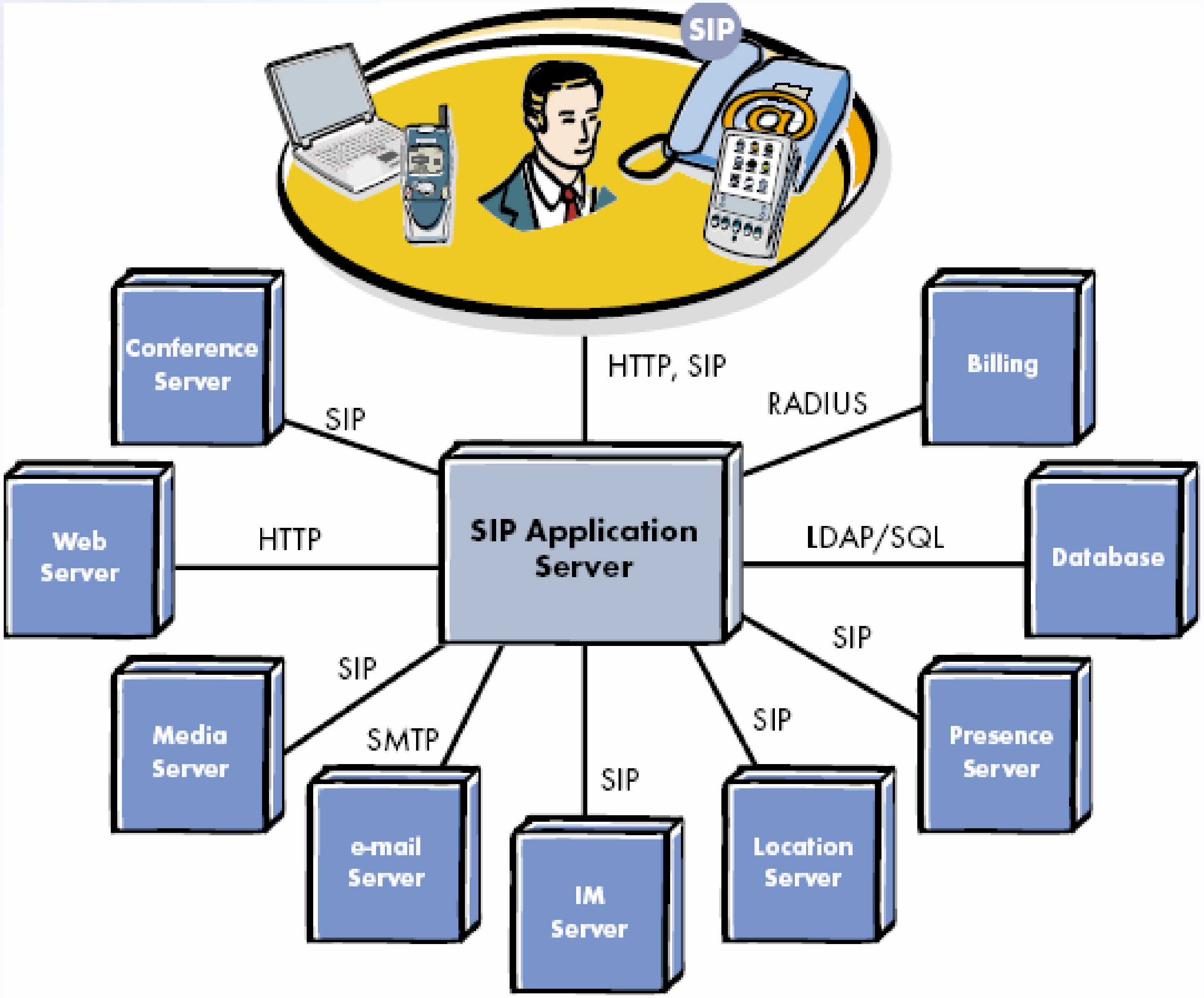
# Elementos do SIP

- **SIP Proxy Server**
  - "Relays call signaling", atua tanto como cliente e servidor
  - Statefull X Stateless
- **SIP Redirect Server**
  - Redireciona as chamadas para outros servidores.
- **SIP Registrar Server**
  - Aceita requisições de registro dos clientes.
  - Mantem informações dos usuários relativo a localização (ex. HLR).

# Arquitetura SIP
## *Proxy Server*



# Funcionalidades Distribuídas



# Comparação SIP x H.323
## *Introdução*

| | H.323 | SIP |
|---|---|---|
| Architecture | Stack | Element |
| Origin | ITU | IETF |
| Transport | Mostly TCP | Mostly UDP |
| Encoding | ASN.1 | HTTP-like |
| Emphasis | Telephony | Multimedia, multicast |
| Address | Aliases | SIP URLs |

# Comparação SIP x H.323
## *Escalabilidade*

H.323

- Interaction between many sub-protocols make it very complex
- Stateful servers in Version 1+2
- H.323v3 more complex

SIP

- SIP and SDP are less complicated
- Servers can be stateless

# Comparação SIP x H.323
## *Facilidade de Implementação*

**H.323**

- H.323 messages are binary
- Encoded using ASN.1
- Special parsers needed to map into readable form and vice versa
- Implementation and debugging complicated

**SIP**

- SIP messages are text-based (unicode supported)
- Easy implemented in Perl, Tcl, Java
- Easy debugging: tcpdump, ngrep, netcat, ...

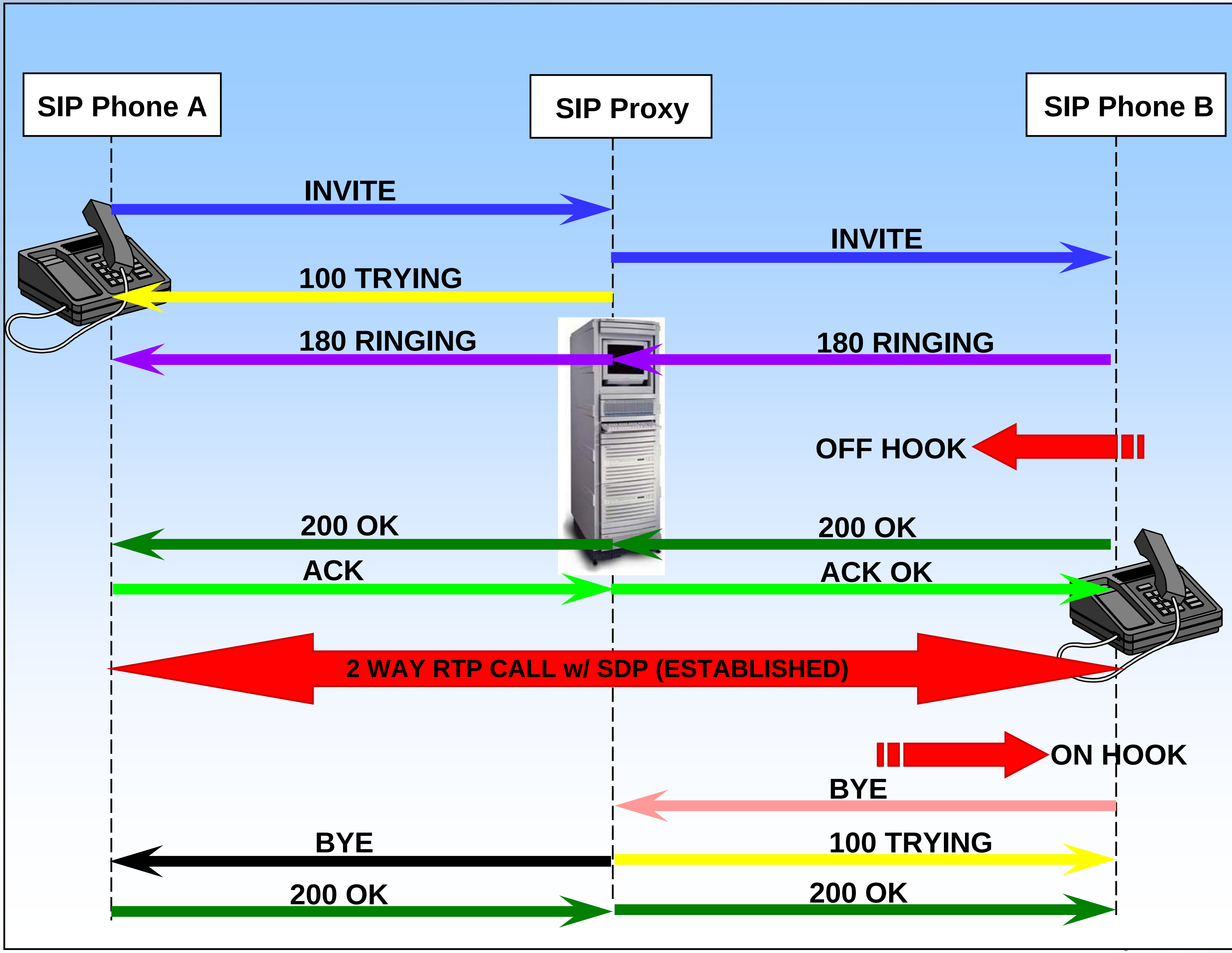
# Comparação SIP x H.323
## *Sumário*

**H.323**

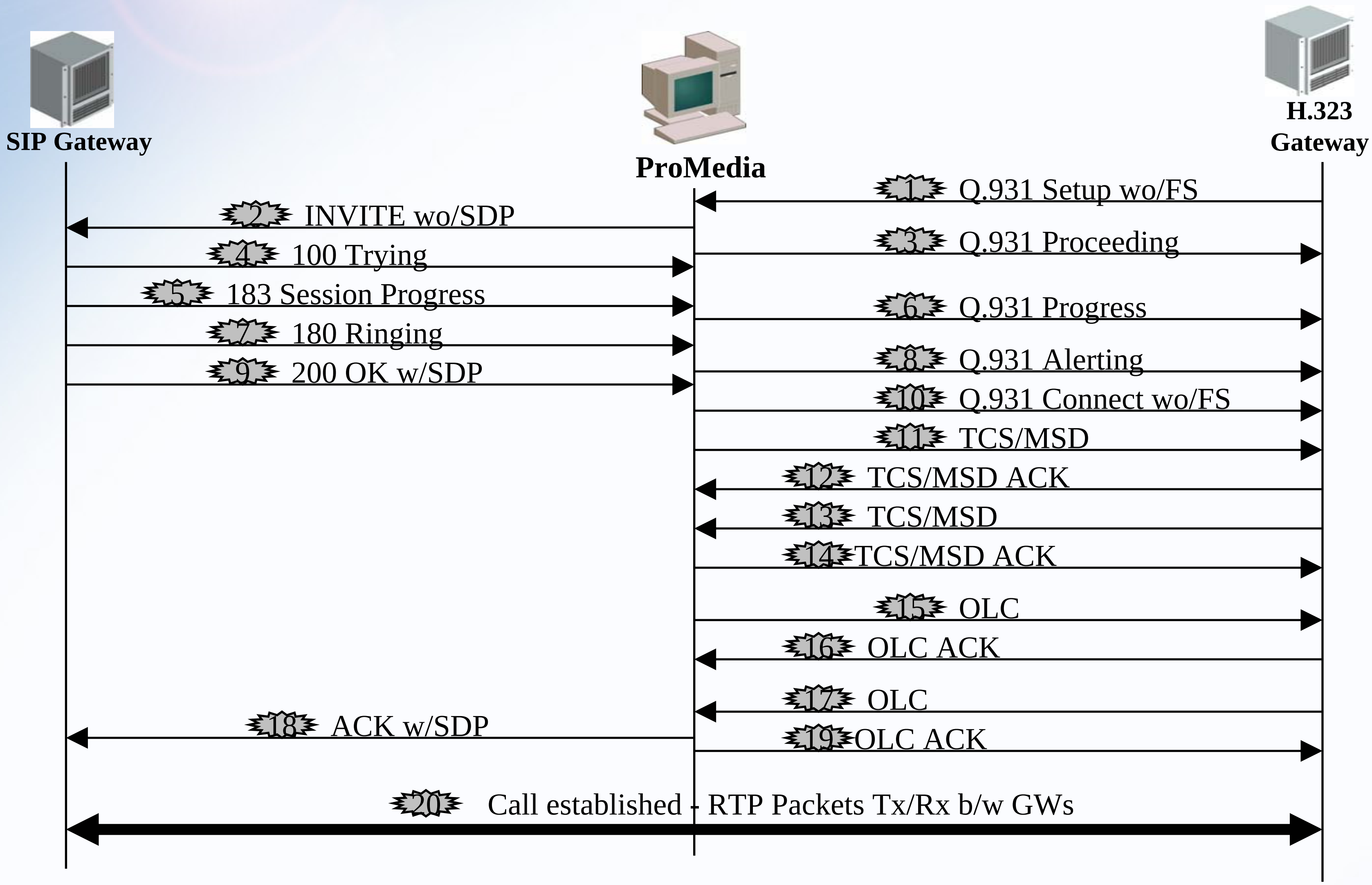
- Deployment started earlier
- Shorter messages

**SIP**

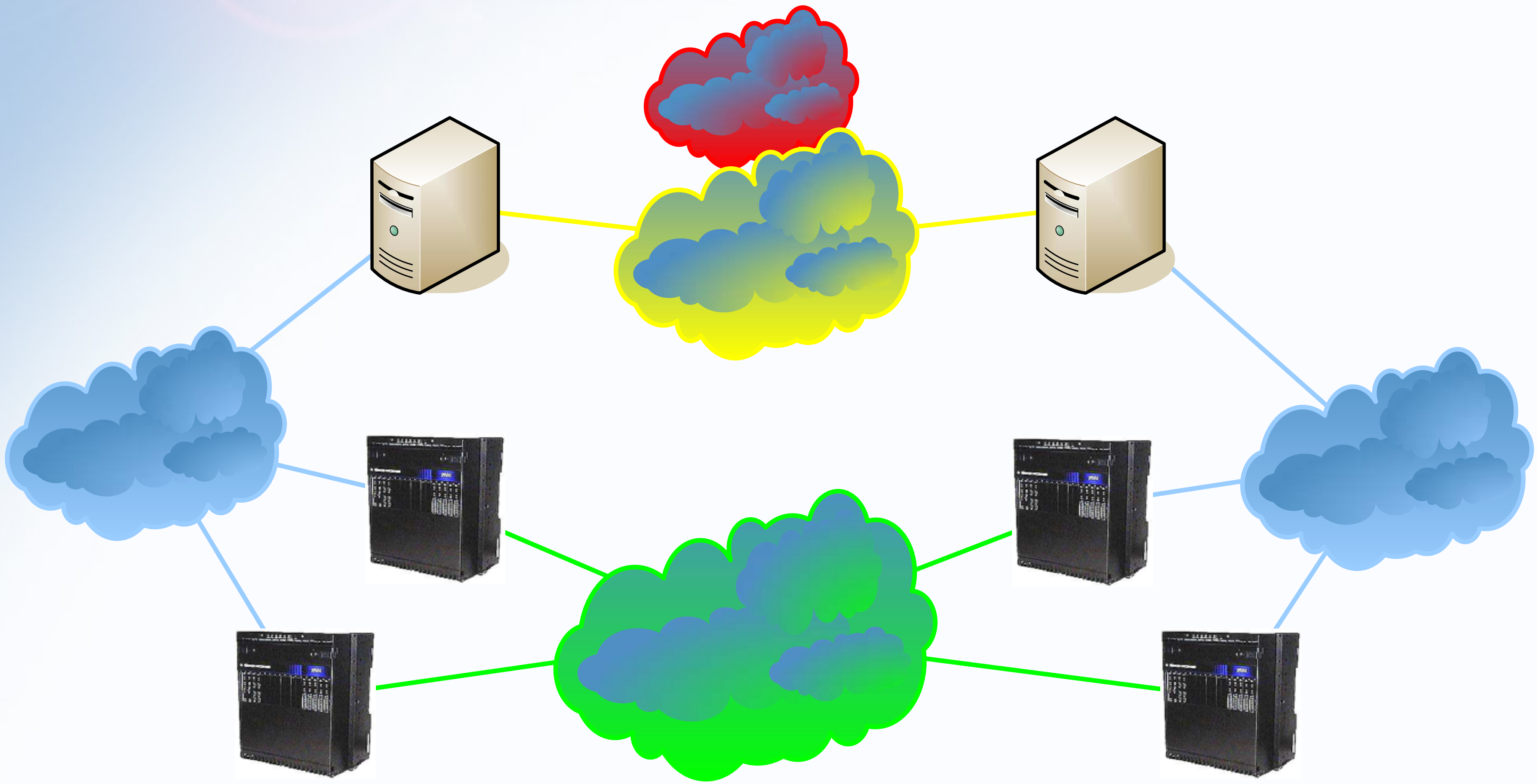
- Scalability
- Extensibility
- Less Complexity
- Ease of Implementation
- Customization
- Call forking
- Third-party call control


SIP Phone A
SIP Proxy
SIP Phone B
INVITE
INVITE
100 TRYING
180 RINGING
180 RINGING
OFF HOOK
200 OK
200 OK
ACK
ACK OK
2 WAY RTP CALL w/ SDP (ESTABLISHED)
ON HOOK
BYE
BYE
100 TRYING
200 OK
200 OK

# Interoperando SIP e H.323
# *Protocol Mediator*

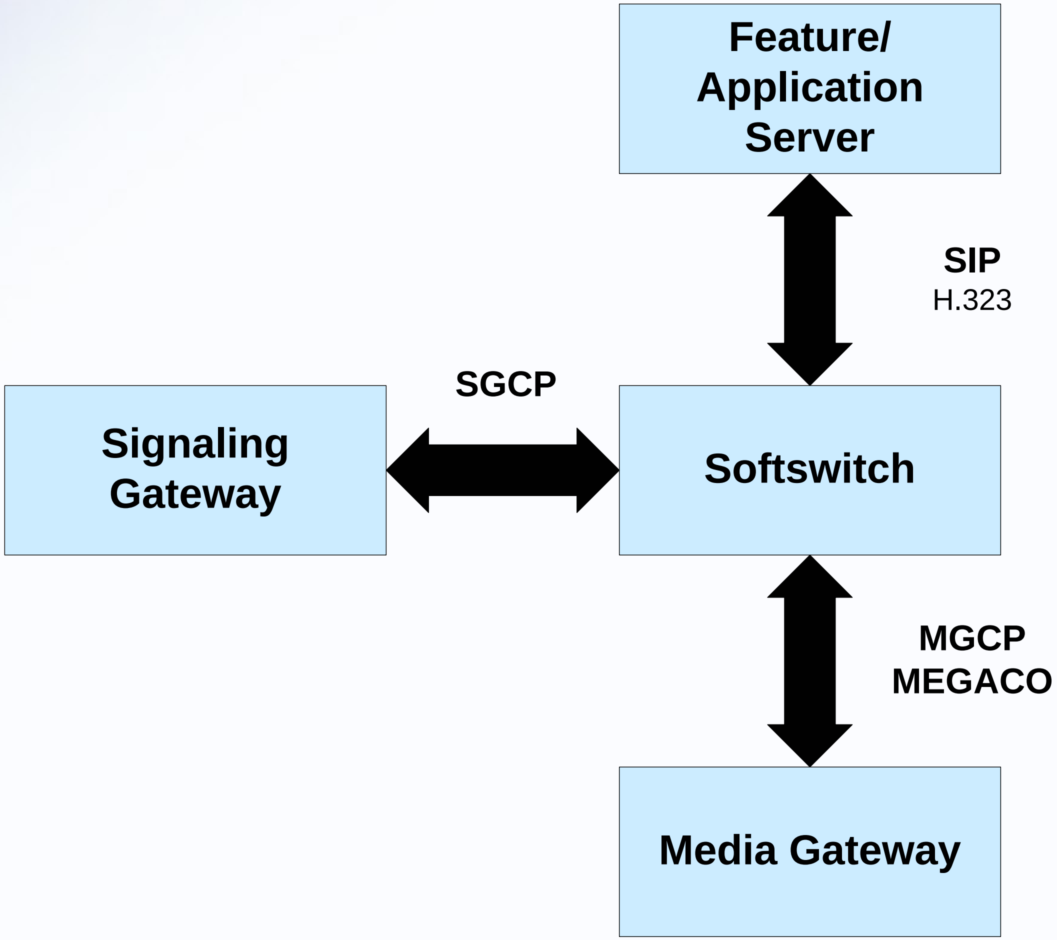


# Protocolos MGCP e MEGACO

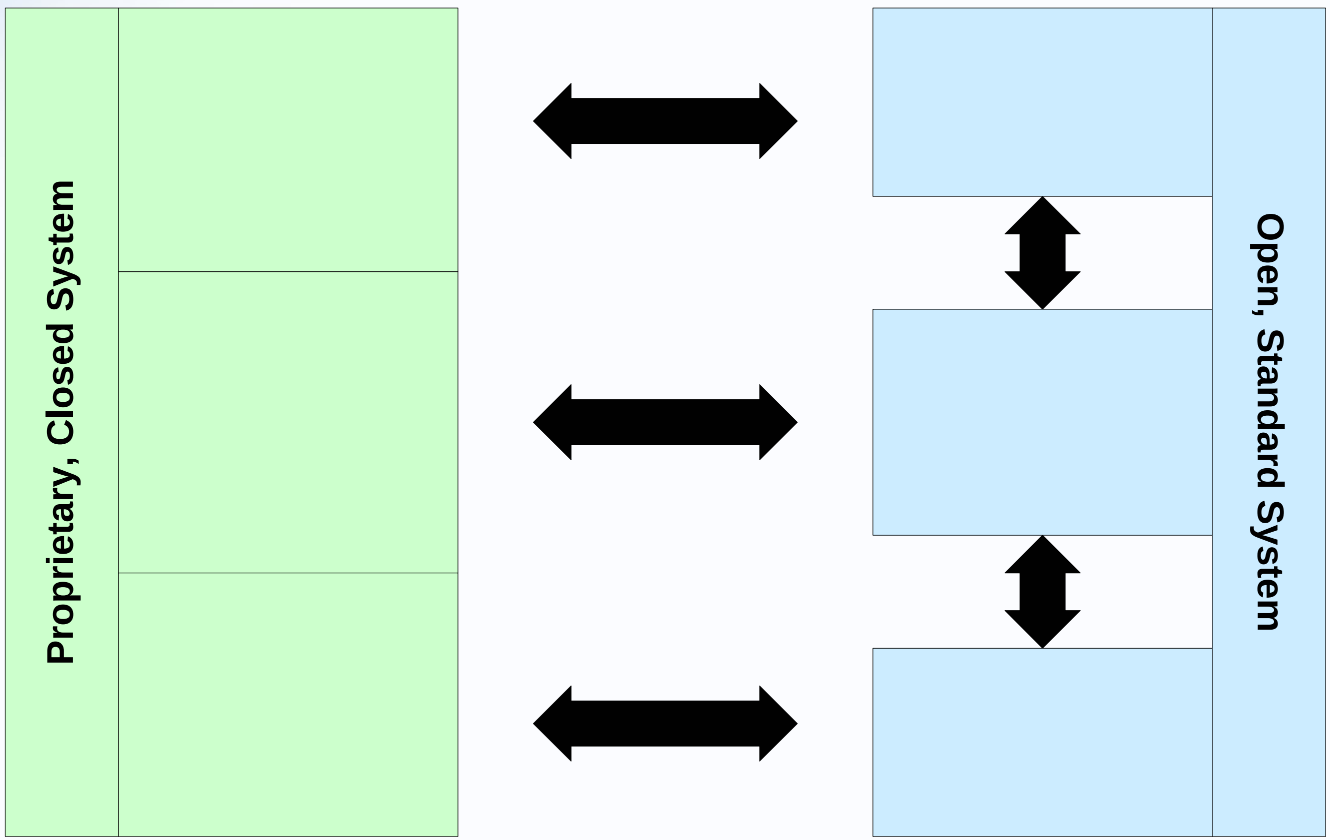


# *O Modelo de Softswitch*

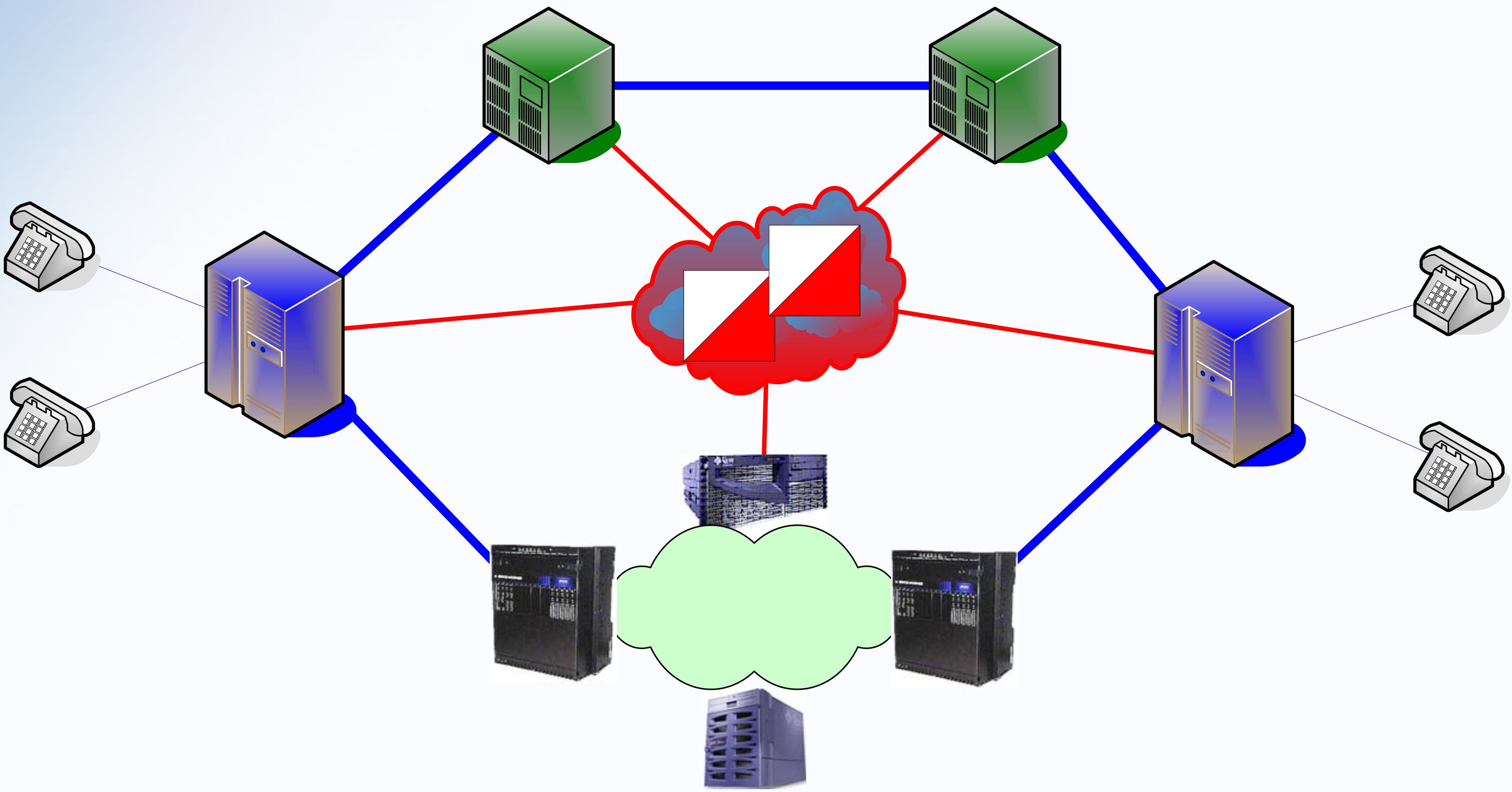
# Arquitetura do Softswitch



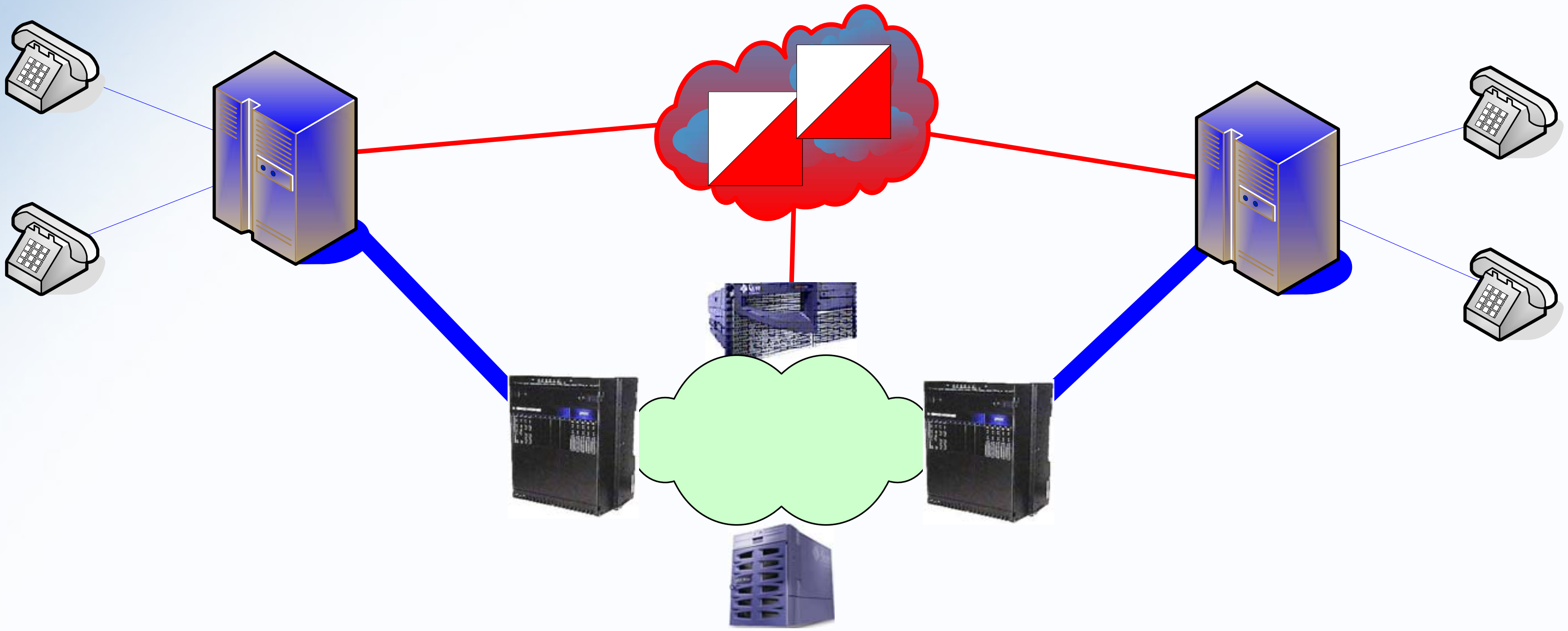
# Arquitetura Softswitch
# X
# Arquitetura Proprietária



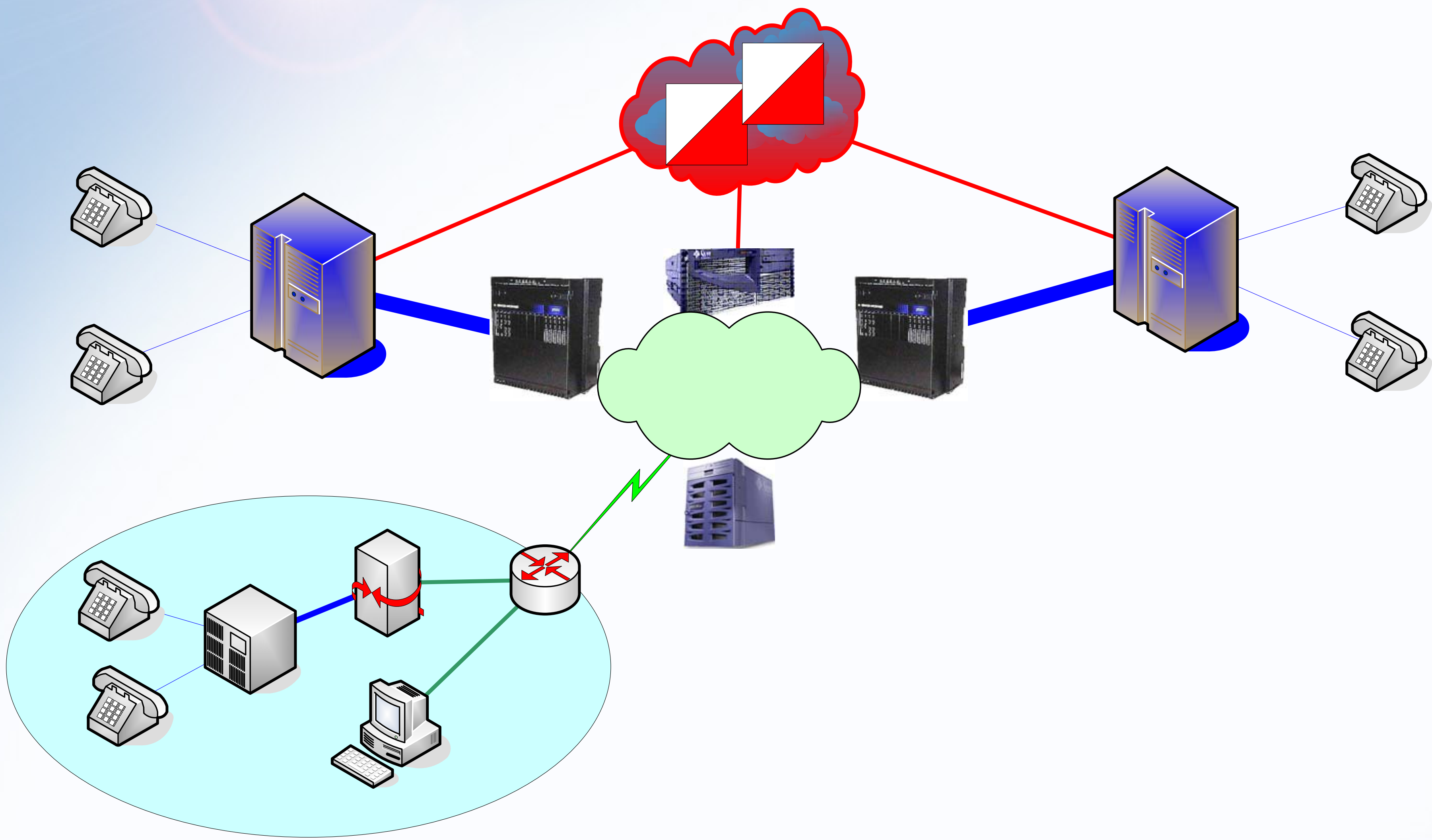
# Implementações de Softswitch



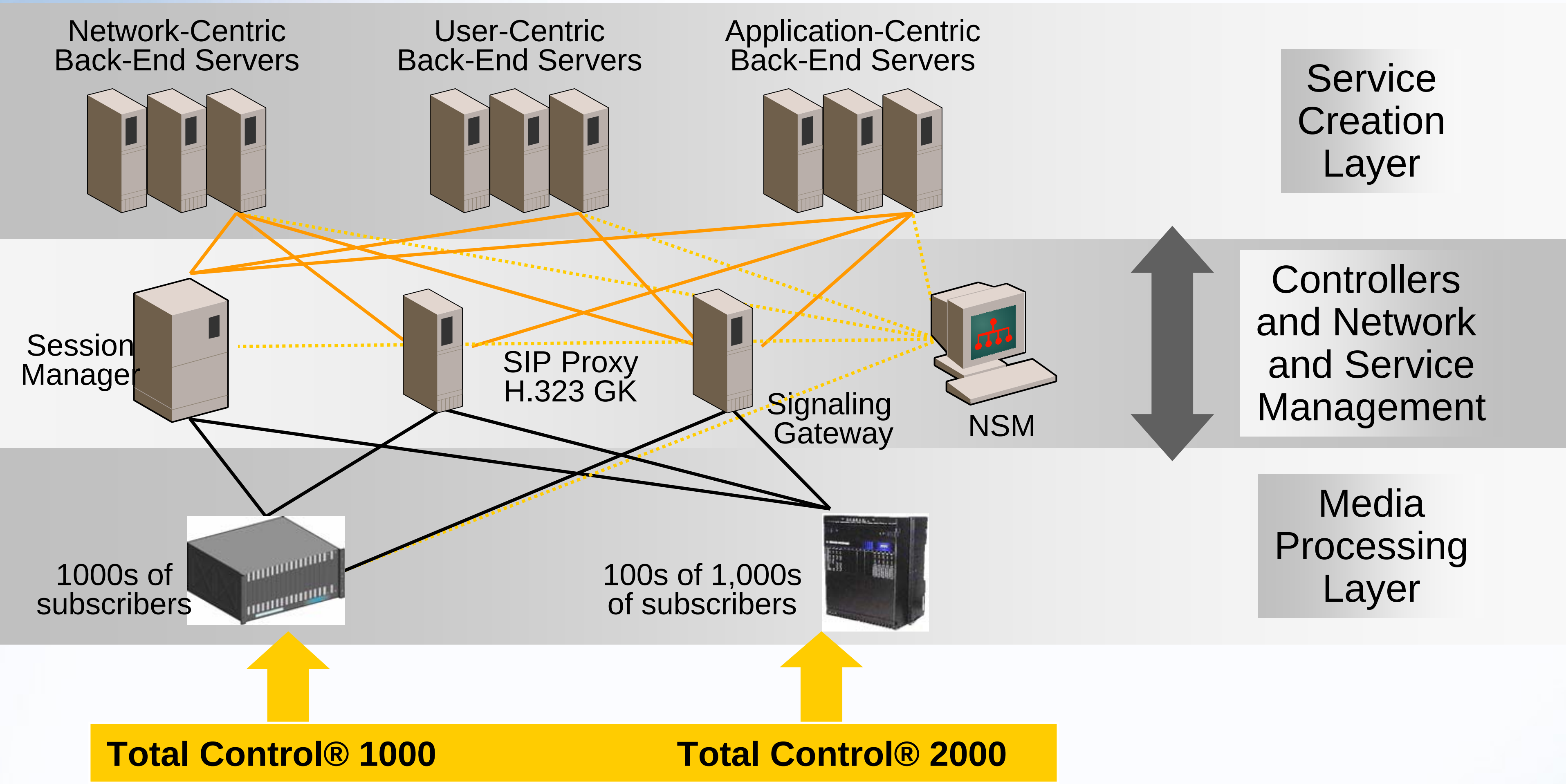
# Implementações de Softswitch

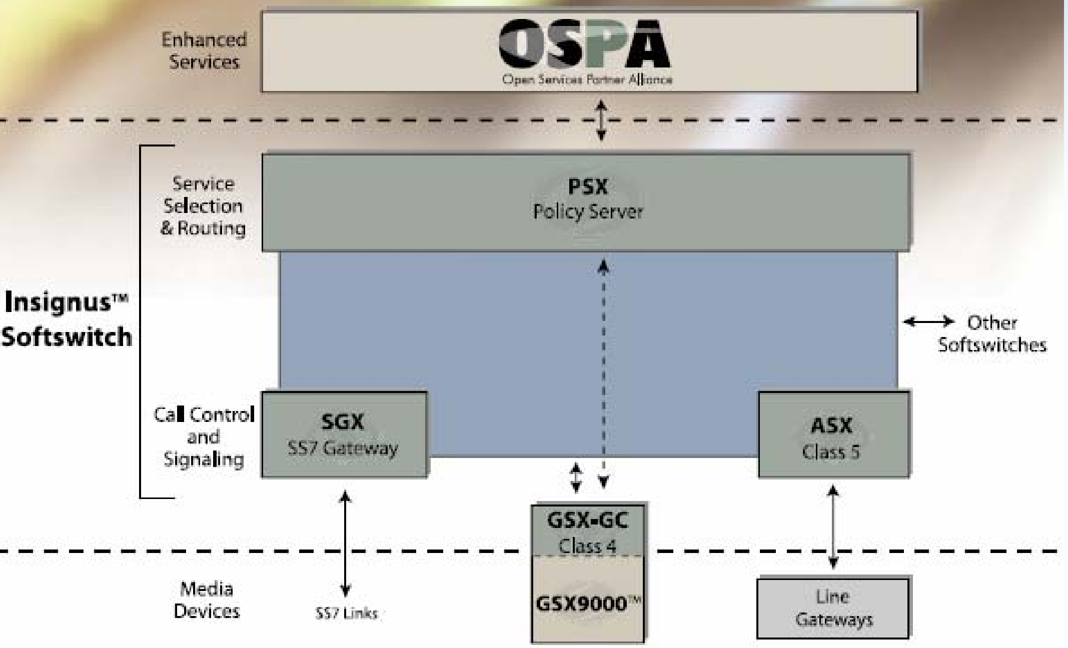


# Implementações de Softswitch

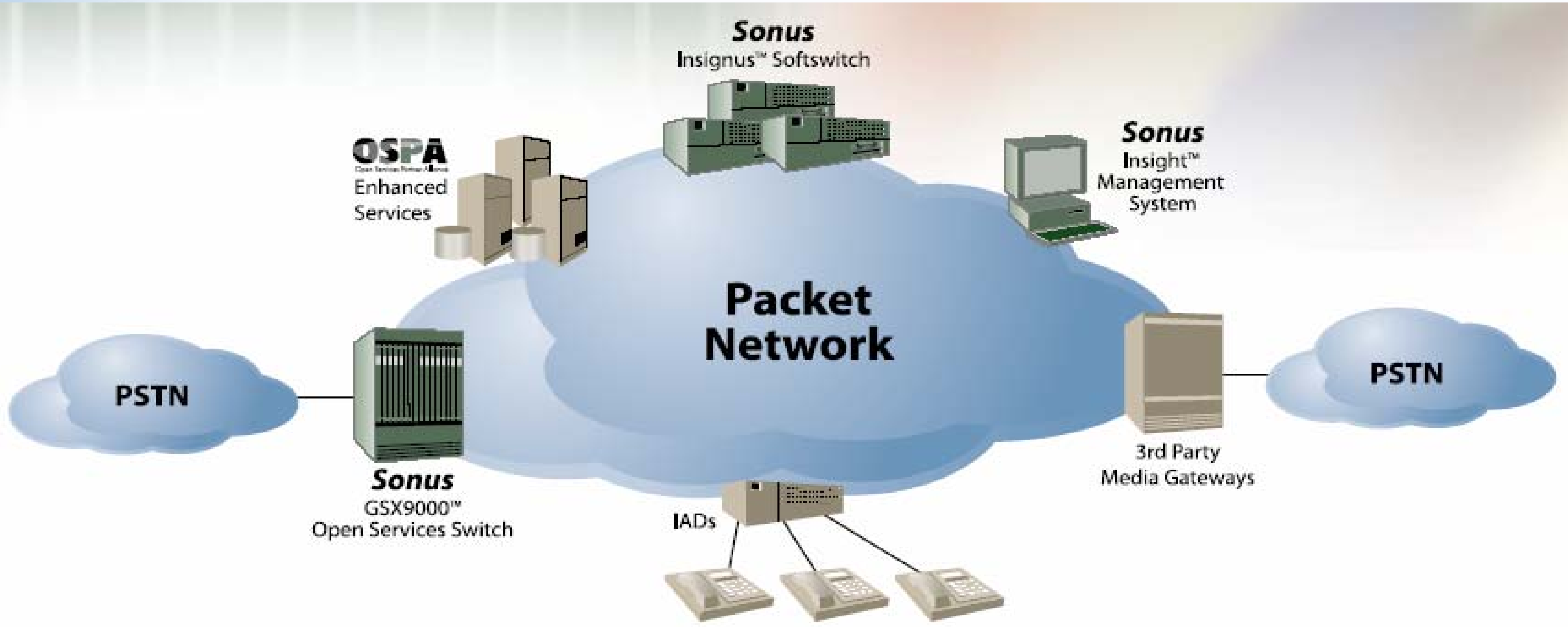


# CommWorks® Softswitch



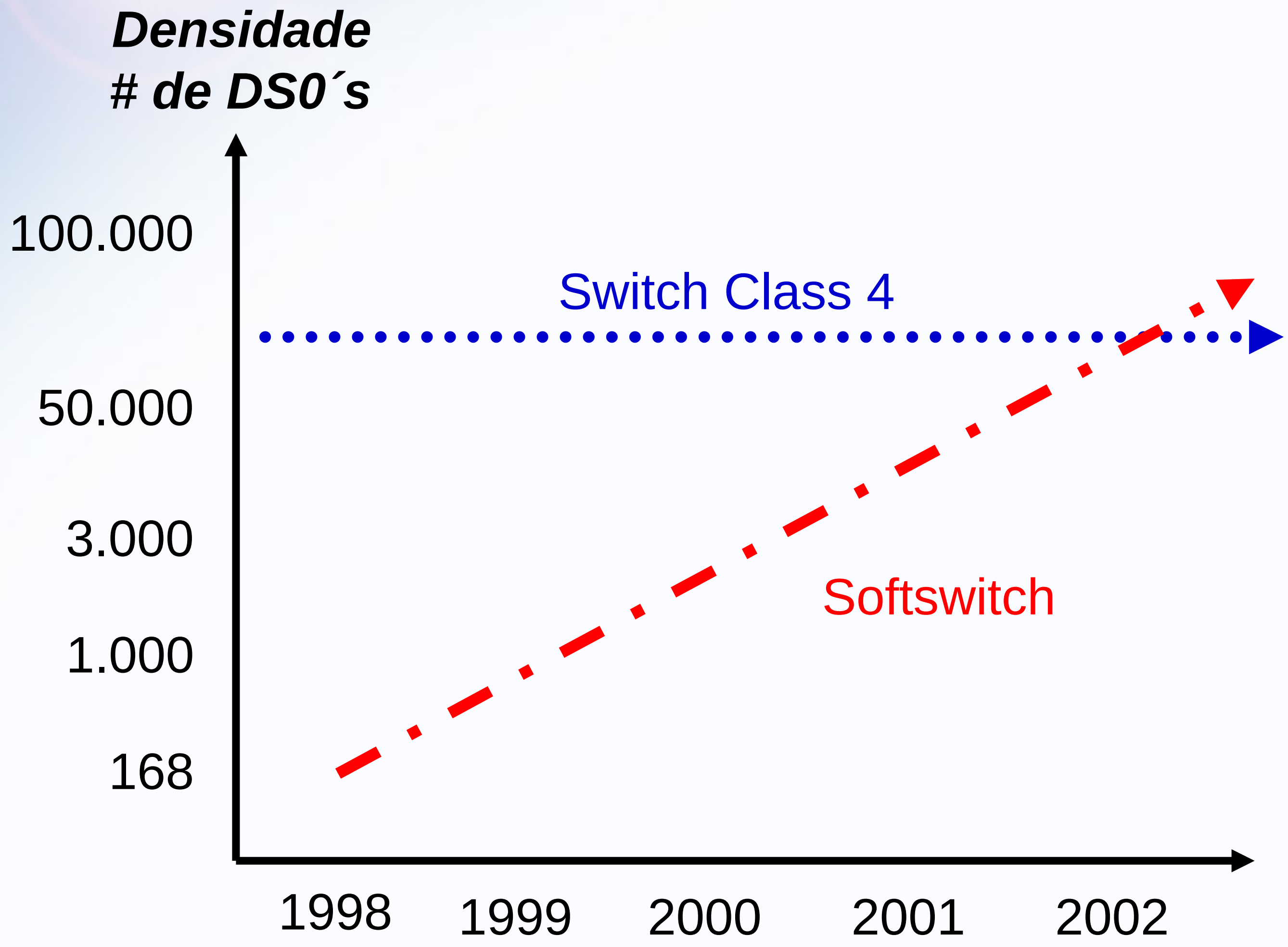



# Sonus® Softswitch

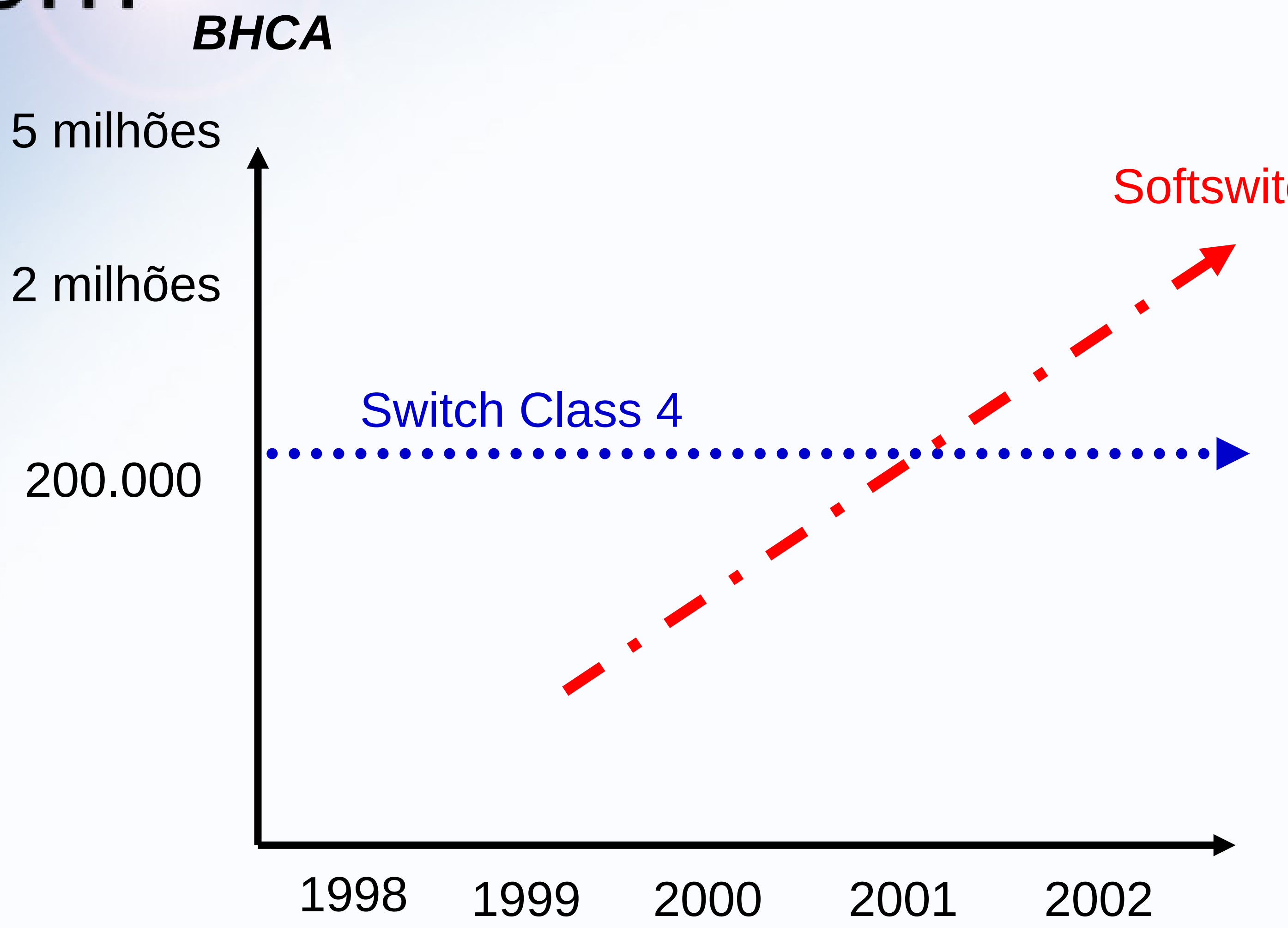


# Softswitch X Class 4&5 Switches

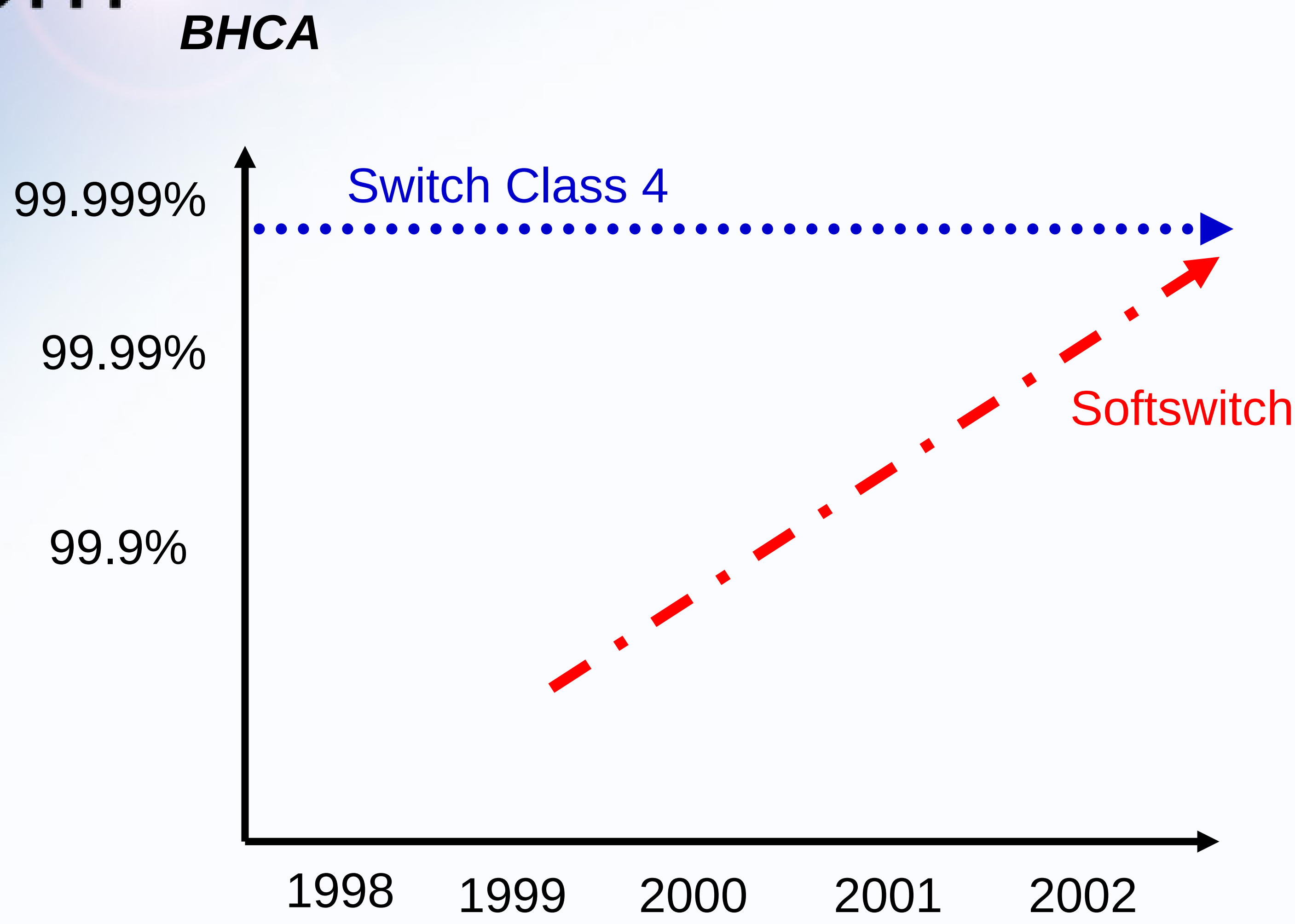
# Circuit Switch X Softswitch
## *Escalabilidade*



# Circuit Switch X Softswitch
## *Escalabilidade*



# Circuit Switch X Softswitch
## *Confiabilidade*

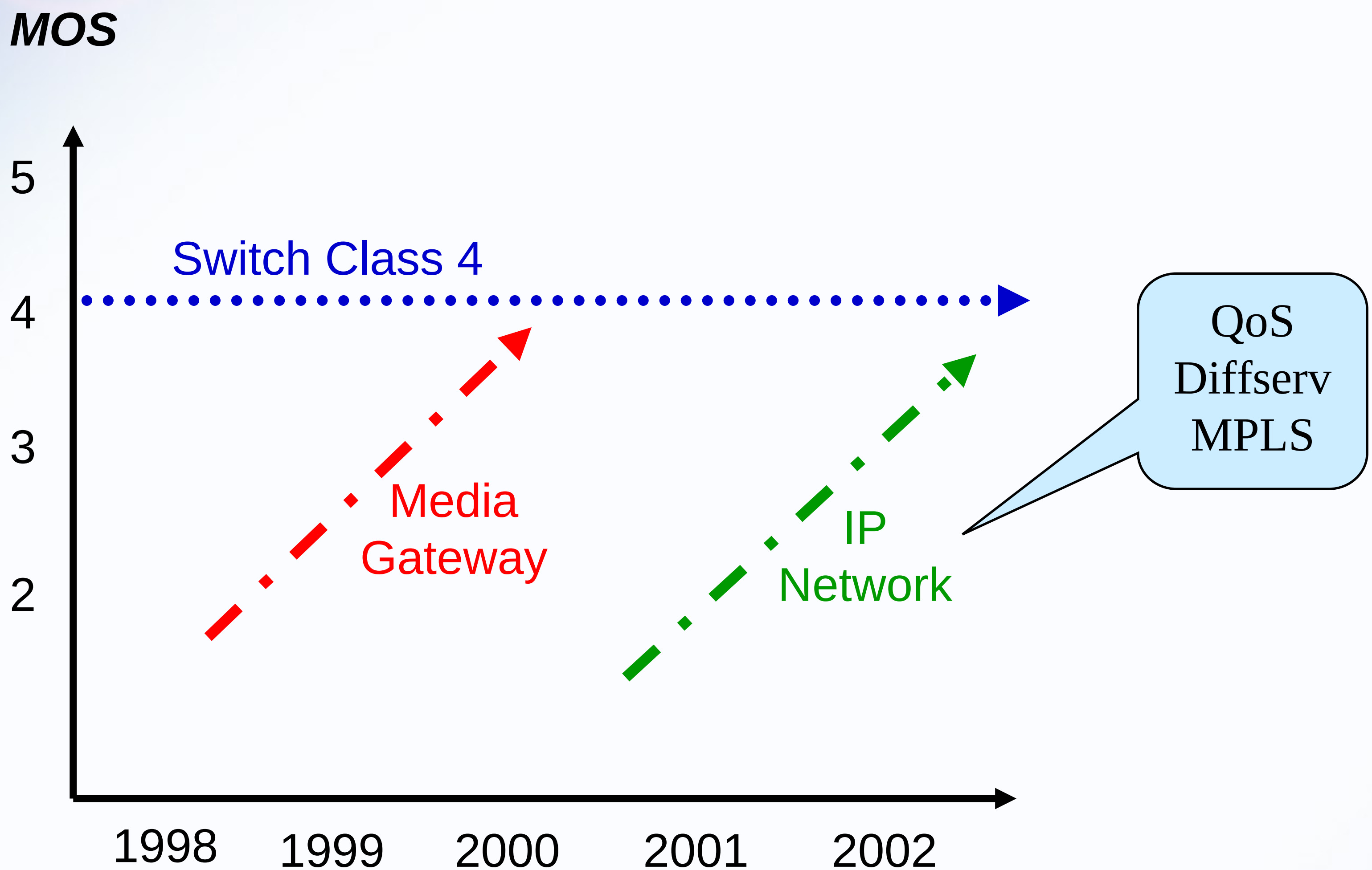


# Comparativo Class4 x Softswitch
## *Exemplos*

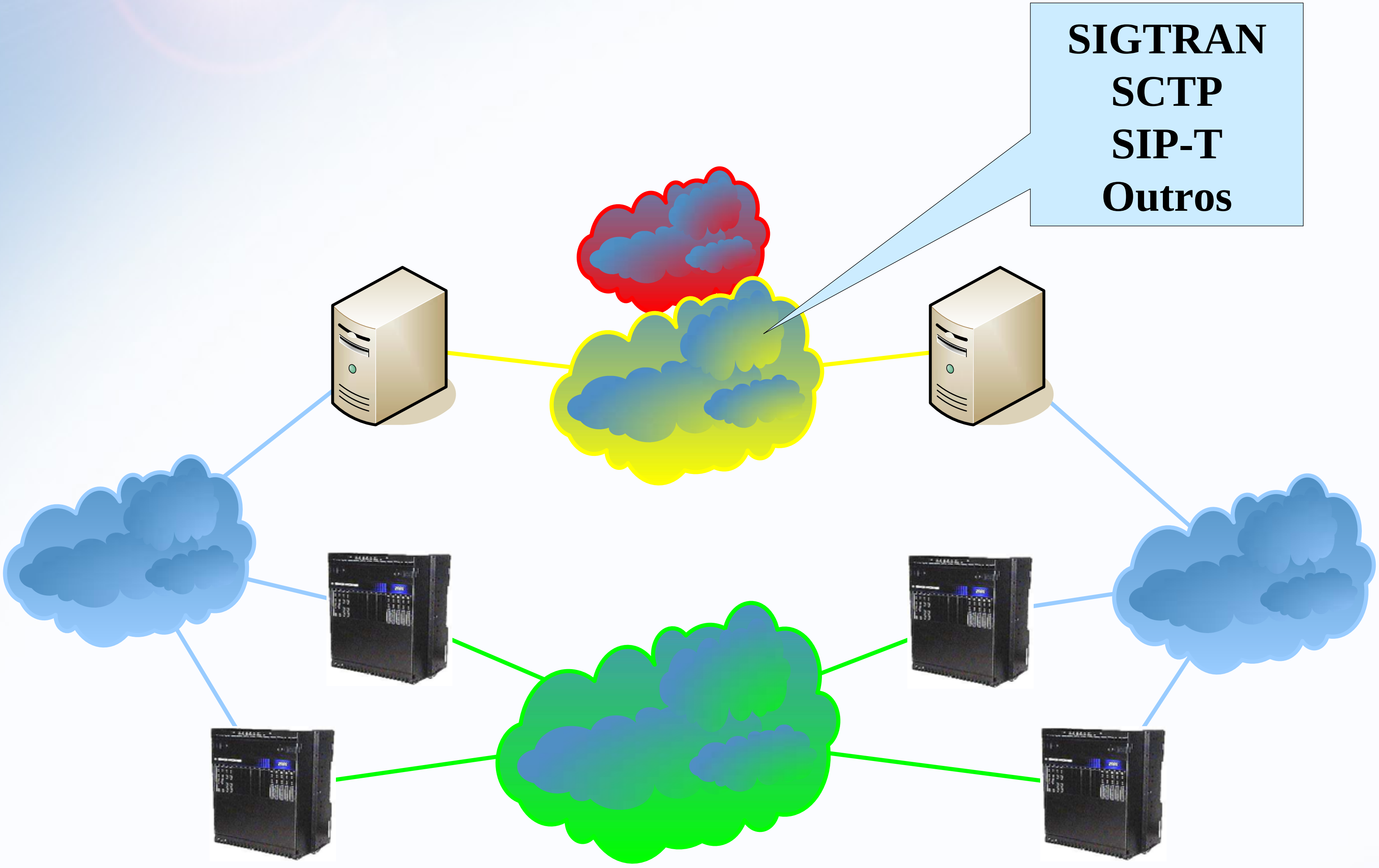
| ***Switch Class 4*** | ***DSO´s*** | ***BHCA´s*** | ***Reliability*** | ***NEBS3*** | ***Price(DS0)*** |
|---|---|---|---|---|---|
| Nortel DMS-250 | 2.688 | 800.000 | 99.999 | Y | $100 |
| Lucent 4ESS | 2,688 | 700.000 | 99.999 | Y | $100 |

| ***Softswitch*** | ***DSO´s*** | ***BHCA´s*** | ***Reliability*** | ***NEBS3*** | ***Price(DS0)*** |
|---|---|---|---|---|---|
| Convergent ICS | 108.864 | 1,5Mi | 99.999 | Y | $25 |
| Sonus GSX | 24.000 | 2Mi | 99.999 | Y | $25 |
| Commworks | 50.000 | 1Mi | 99.999 | Y | $25 |

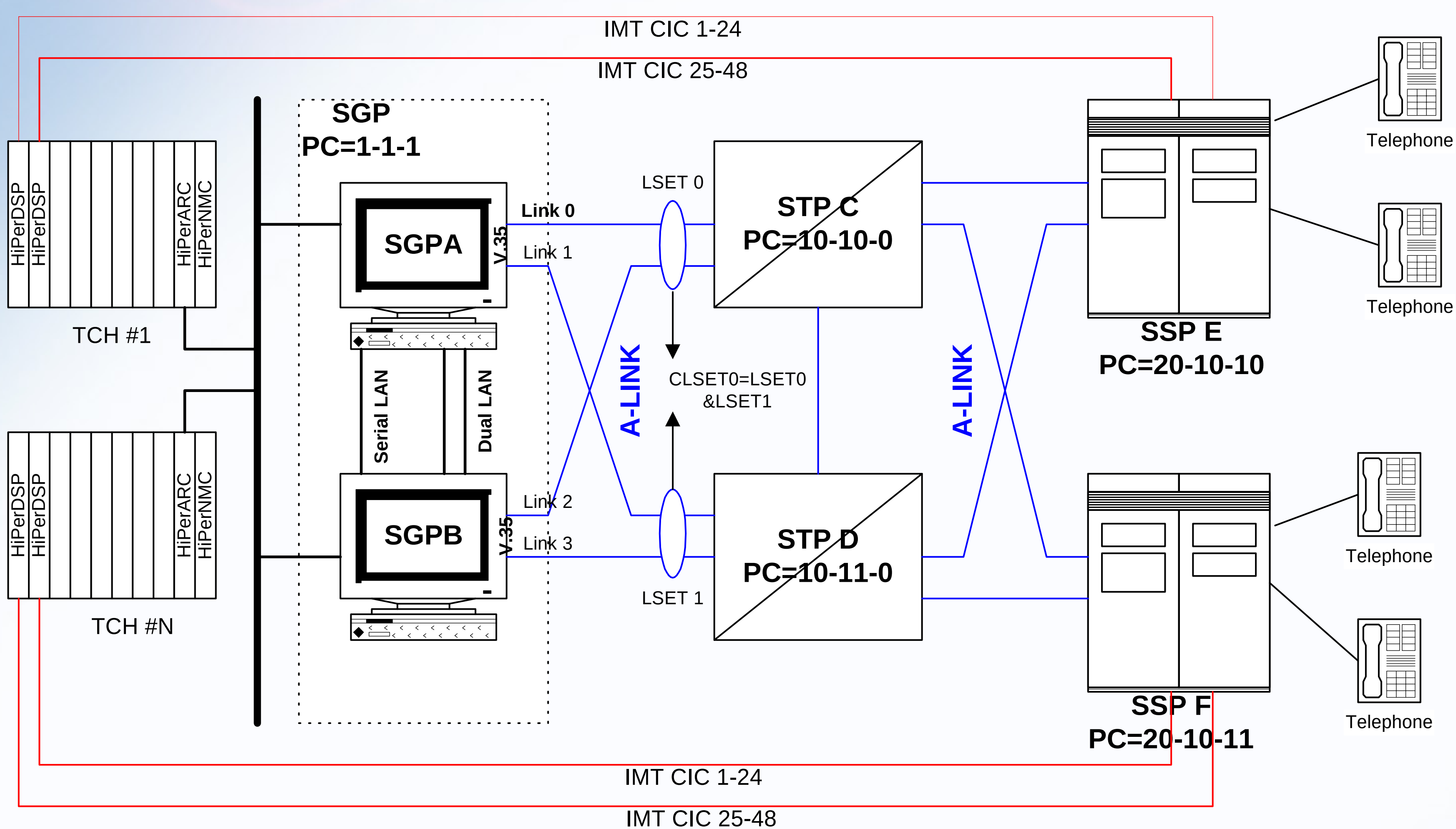
# Circuit Switch X Softswitch
# *Qualidade da Voz*



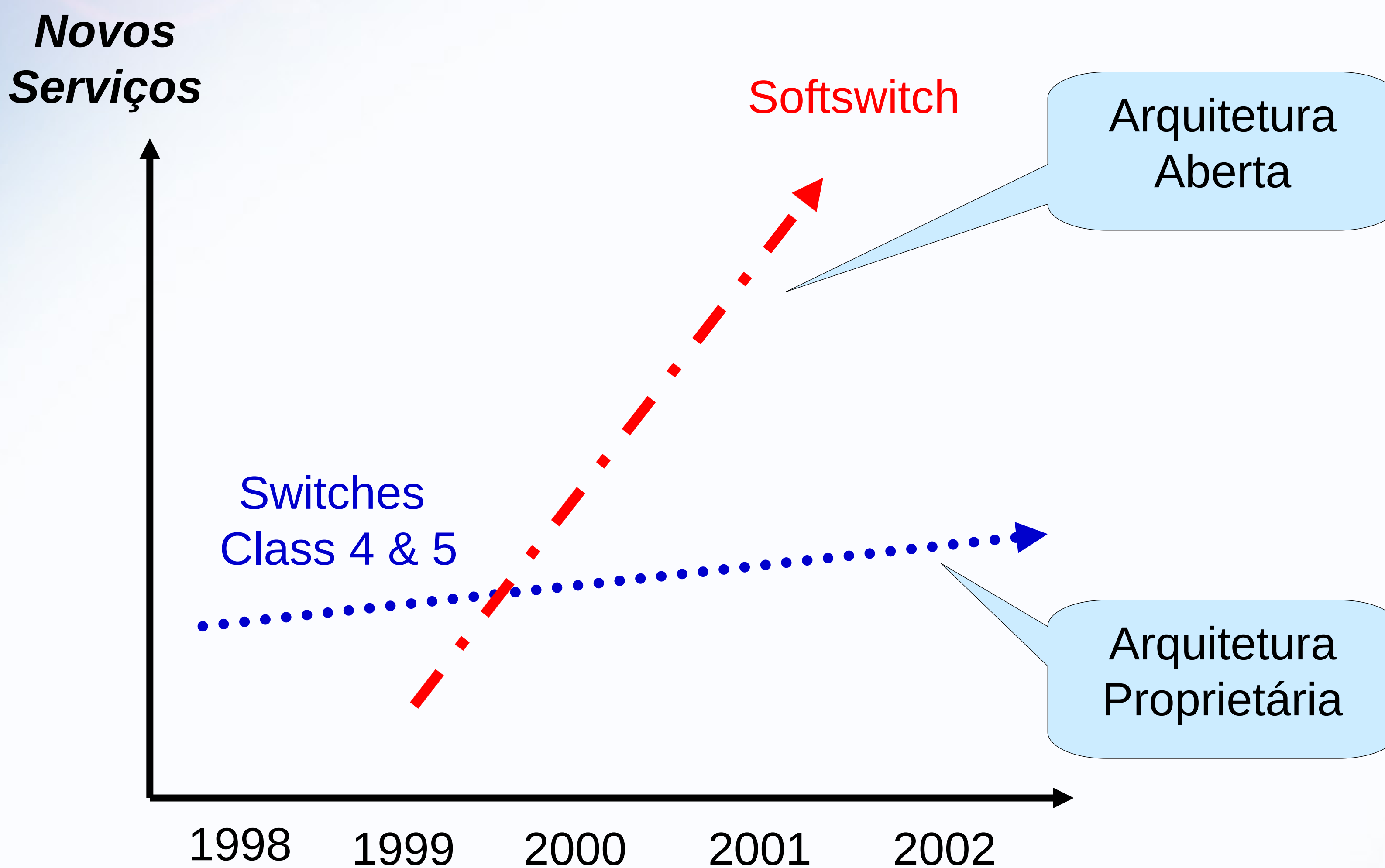
# Circuit Switch X Softswitch
## *Sinalização*



# Circuit Switch X Softswitch
# *Sinalização*

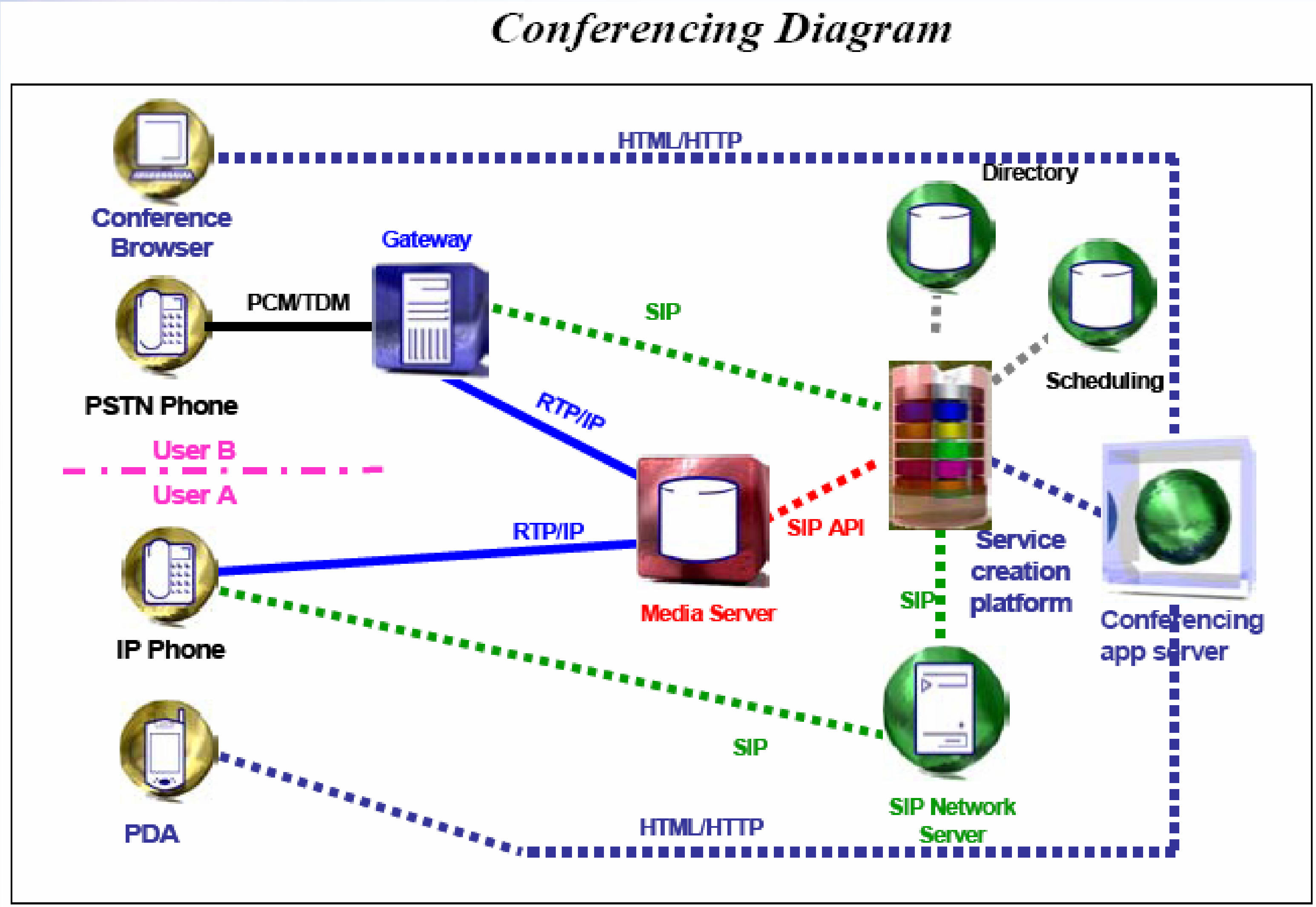


# Circuit Switch X Softswitch
## *Novos Serviços*

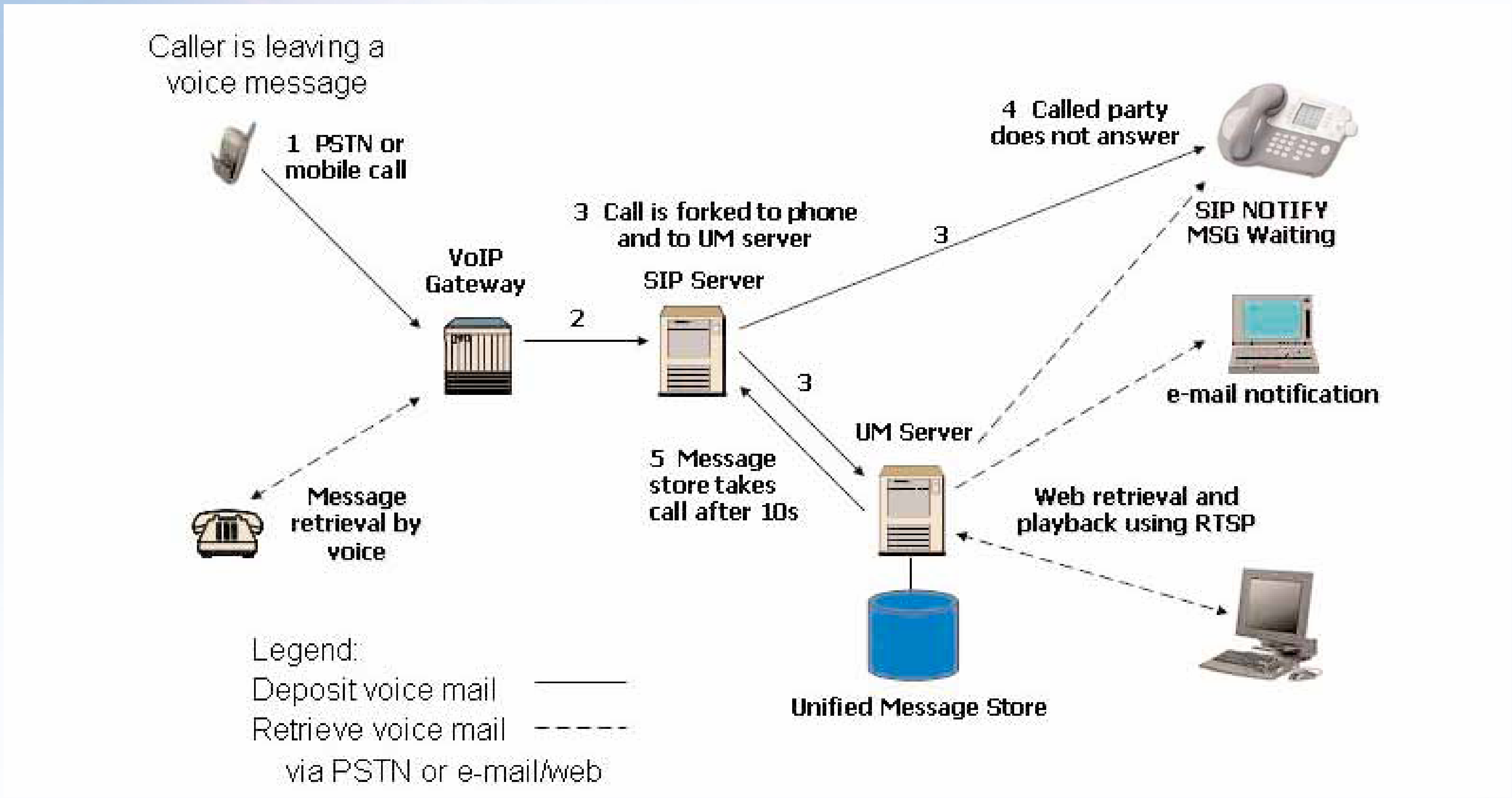


# *Exemplos Aplicações Softswitches*

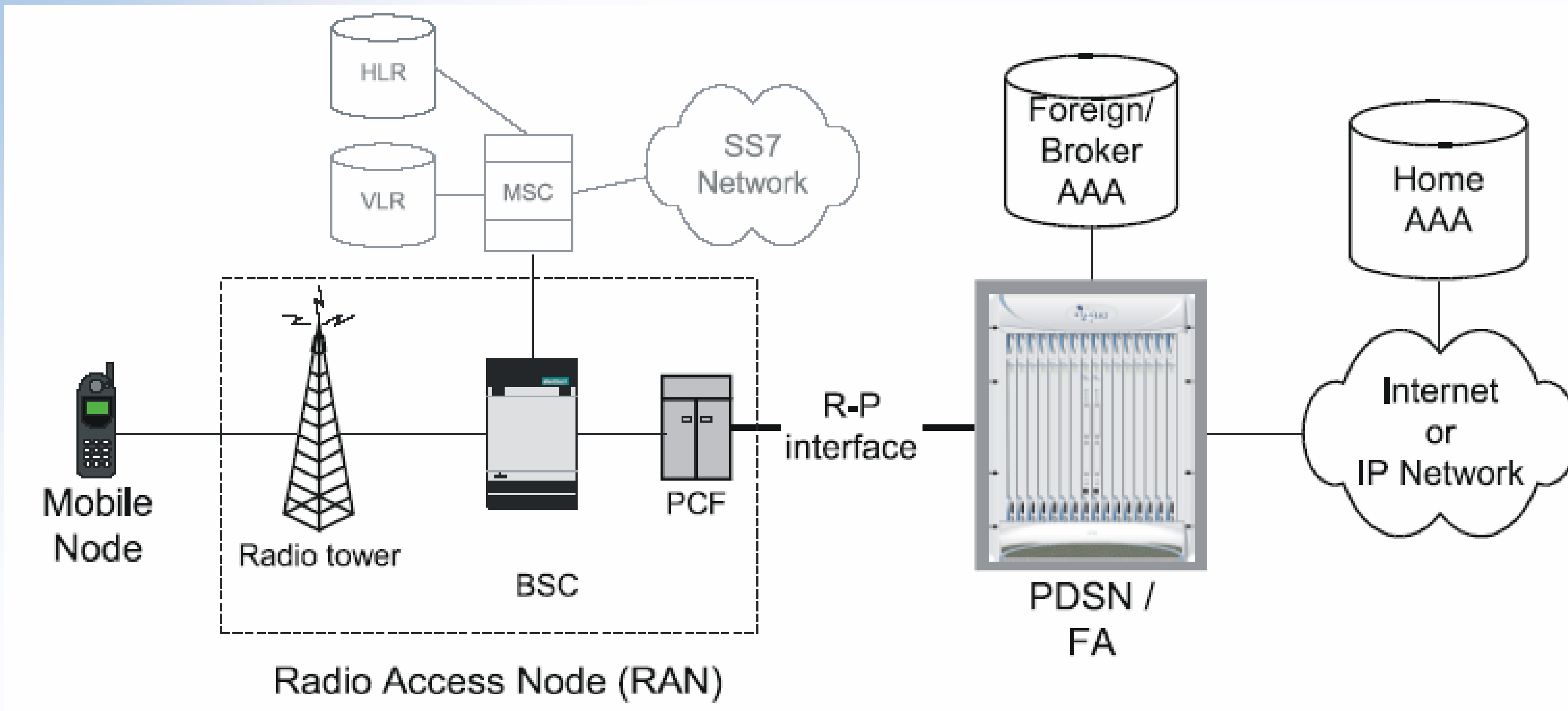
# Conferência entre Dispositivos



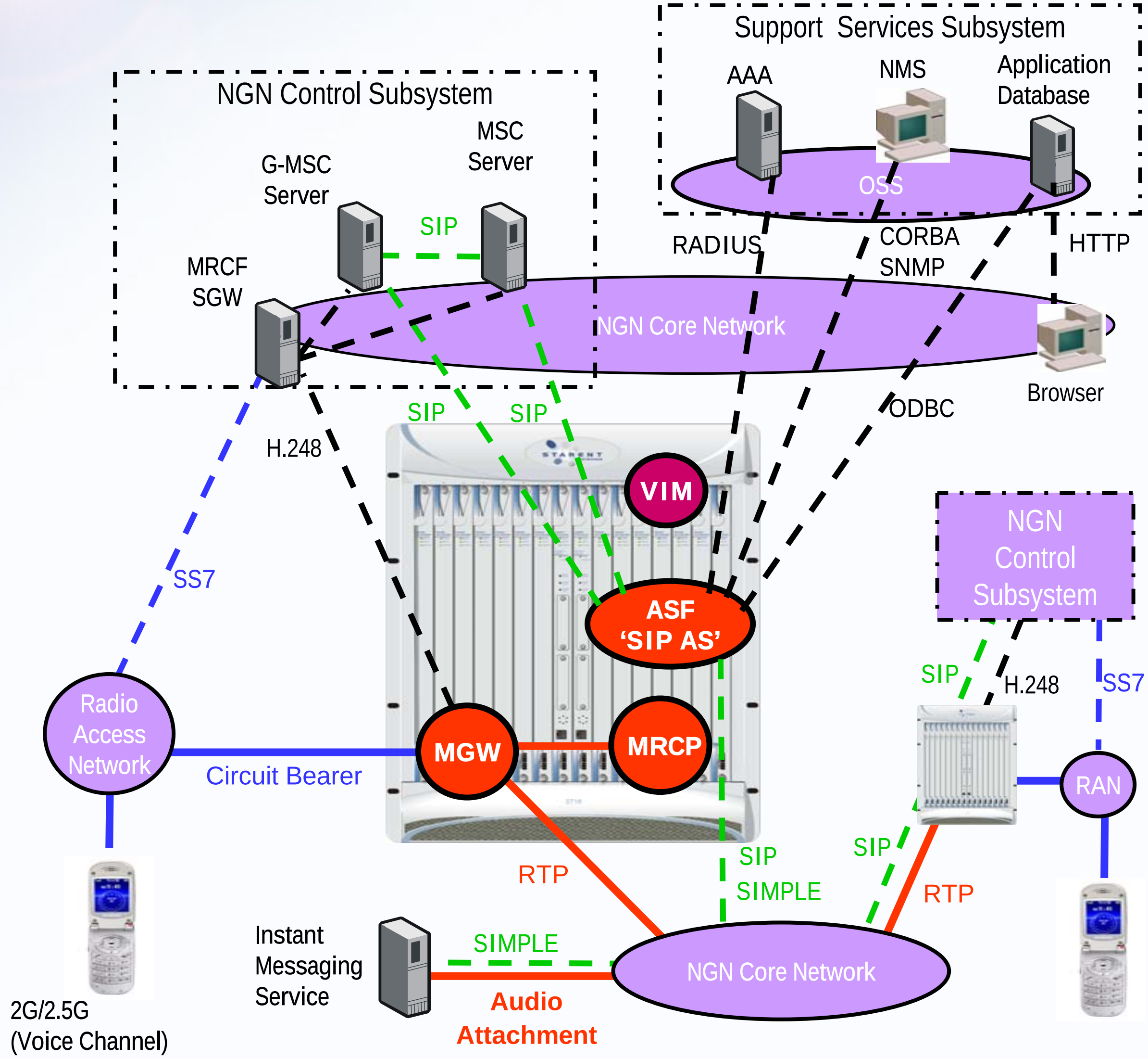
# Unified Messaging



# Voice Instant Messaging
## *Redes 3G*



# Voice Instant Messaging



# *A 3Com e o Softswitch*

# 3Com VoIP Experience

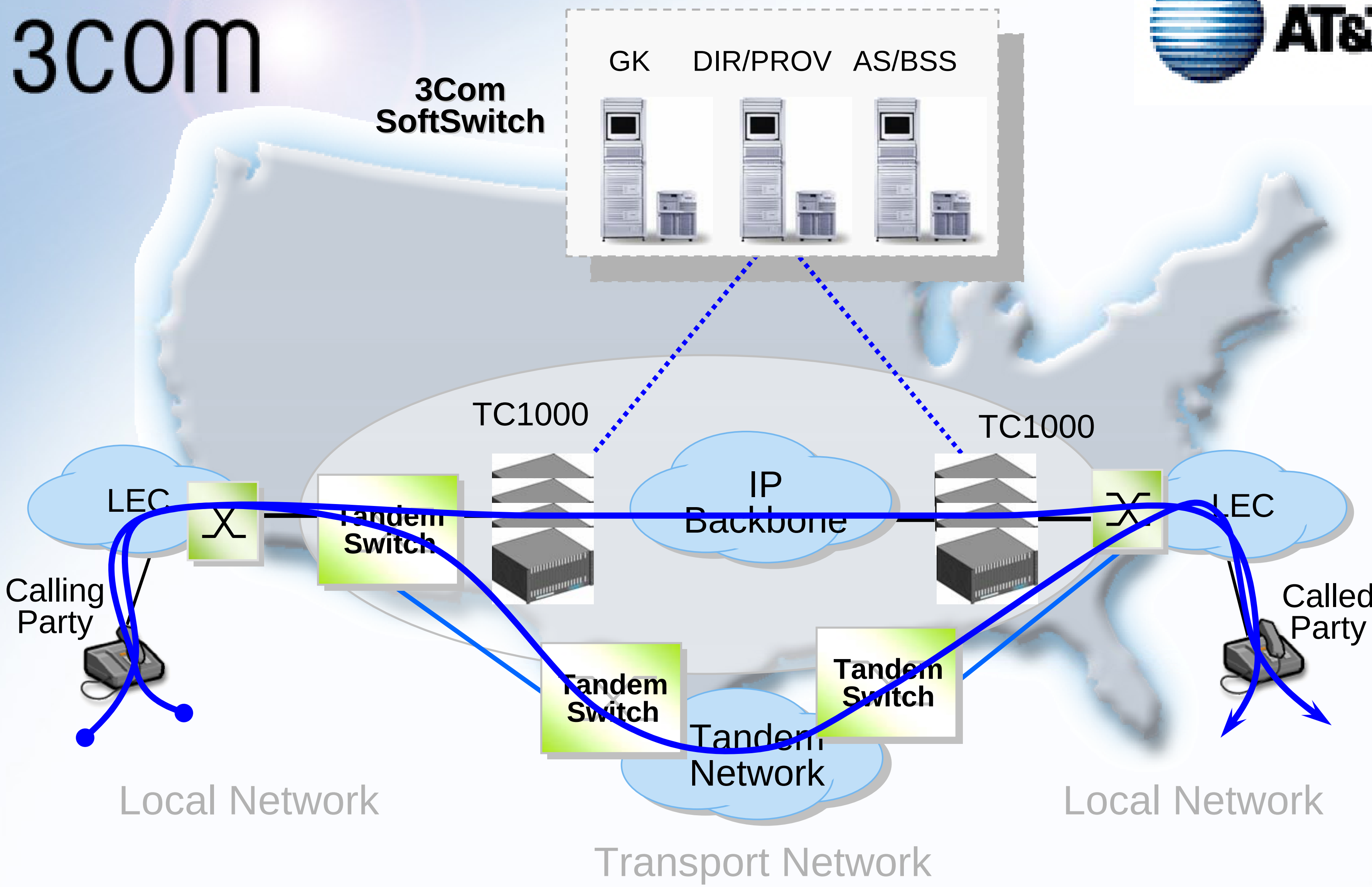
3Com

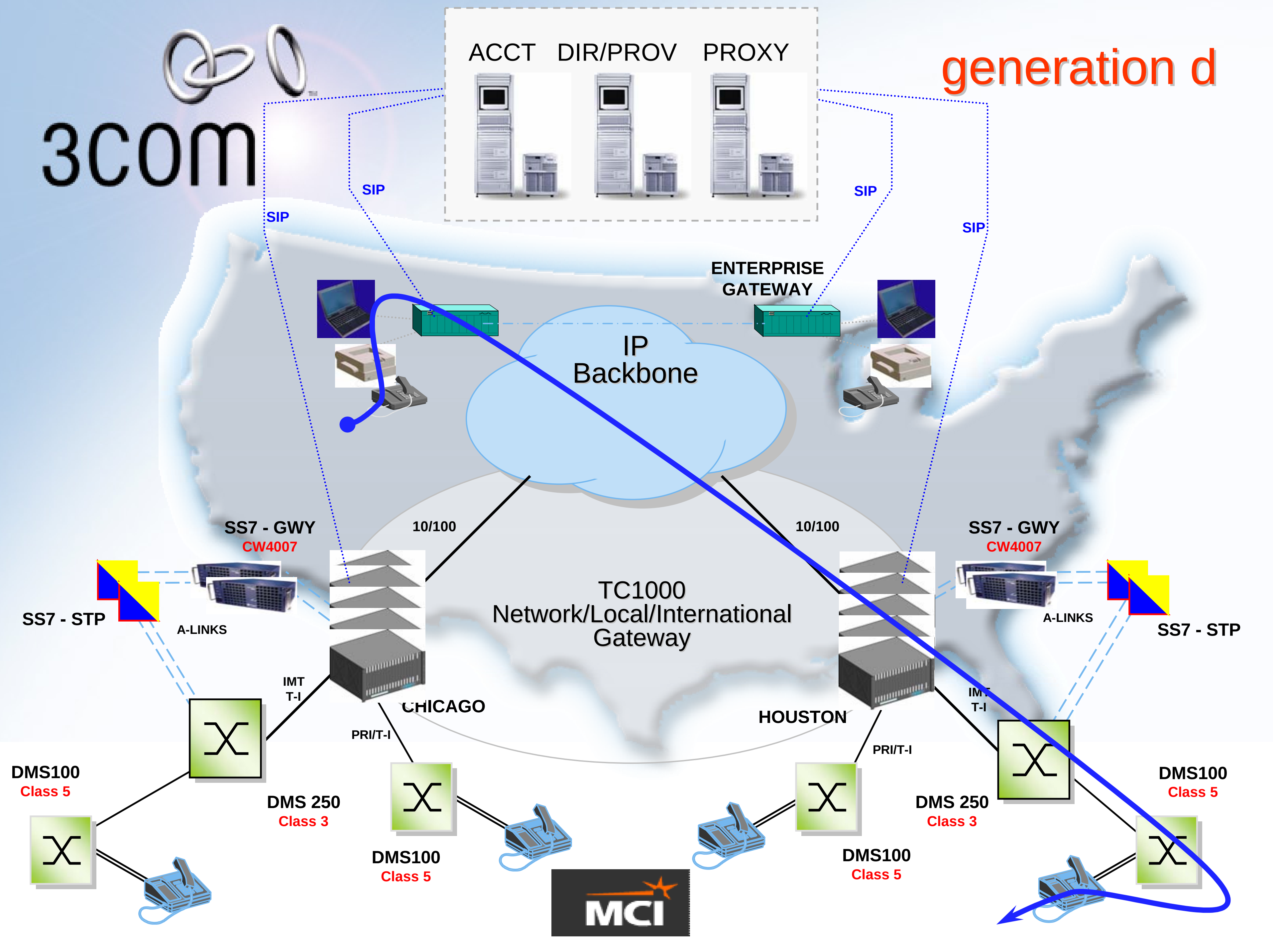
## 3Com Carrier Leadership

- 1998-Entered market with AT&T
- 20 billion minutes carried for SP
- Market leader in softswitch
- Open, standard protocols
- VoIP networks worldwide
  - China Unicom is among the largest in geography (over 320 cities)
  - Reliable in heavy use: AT&T is among the top 3 in minutes of Use (over 13m minutes per day)

## 3Com NBX Leadership

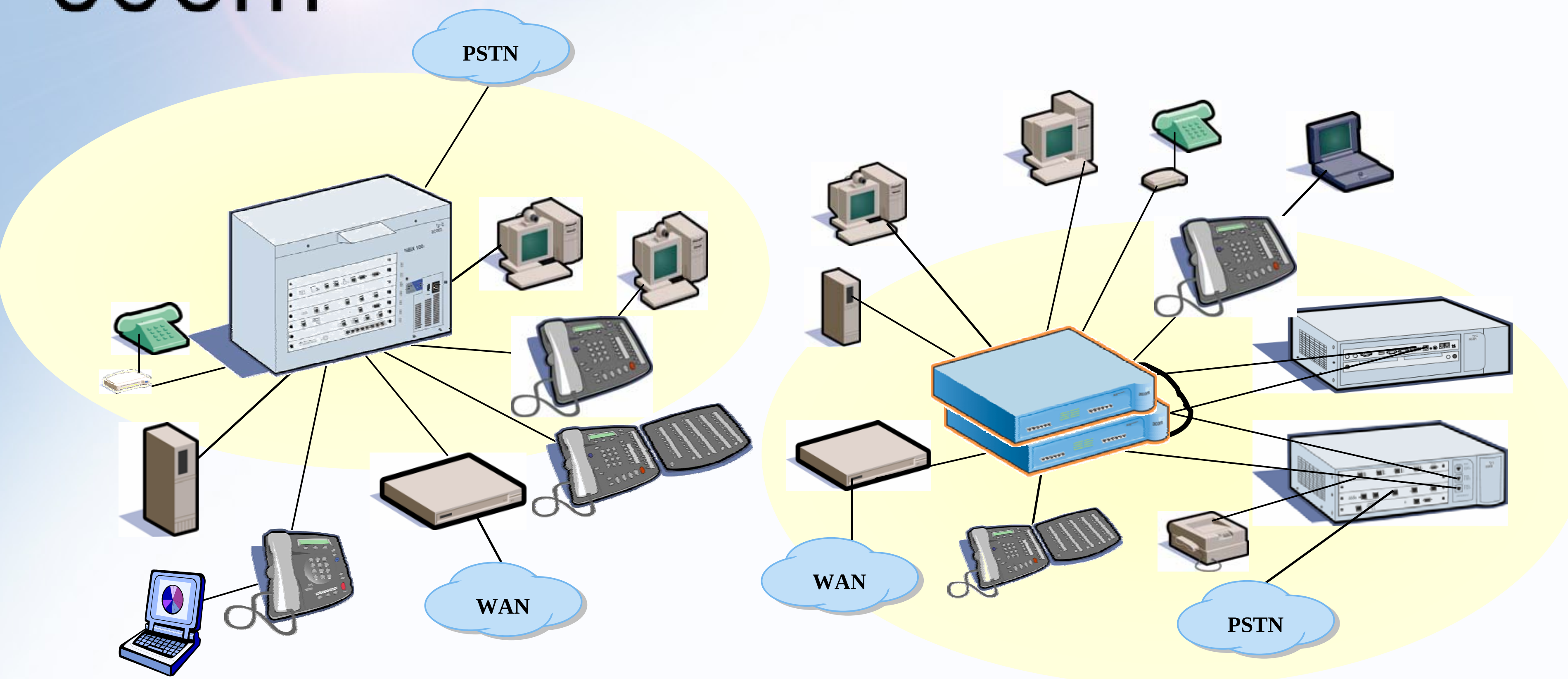
- 1998-1st to market with IP-PBX
- Over 14,500 system deployments
- Over 400,000 phones shipped
- Operates on 3Com & 3rd-party LAN/WAN infrastructure
- Available in over 60 countries
- 40+ industry awards and growing
- #1 or #2 IP-PBX systems share in every quarter since category inception


3Com
SoftSwitch
GK
DIR/PROV
AS/BSS
TC1000
TC1000
IP
Backbone
LEC
LEC
Tandem
Switch
Calling
Party
Called
Party
Tandem
Switch
Tandem
Switch
Tandem
Network
Local Network
Local Network
Transport Network



generation d
3com
ACCT
DIR/PROV
PROXY
SIP
SIP
SIP
SIP
ENTERPRISE GATEWAY
IP Backbone
10/100
10/100
SS7 - GWY
CW4007
SS7 - GWY
CW4007
TC1000 Network/Local/International Gateway
SS7 - STP
A-LINKS
A-LINKS
SS7 - STP
IMT T-I
IMT T-I
CHICAGO
HOUSTON
PRI/T-I
PRI/T-I
DMS100
Class 5
DMS 250
Class 3
DMS100
Class 5
DMS100
Class 5
DMS 250
Class 3
DMS100
Class 5
MCI

# NBX Networked Telephony

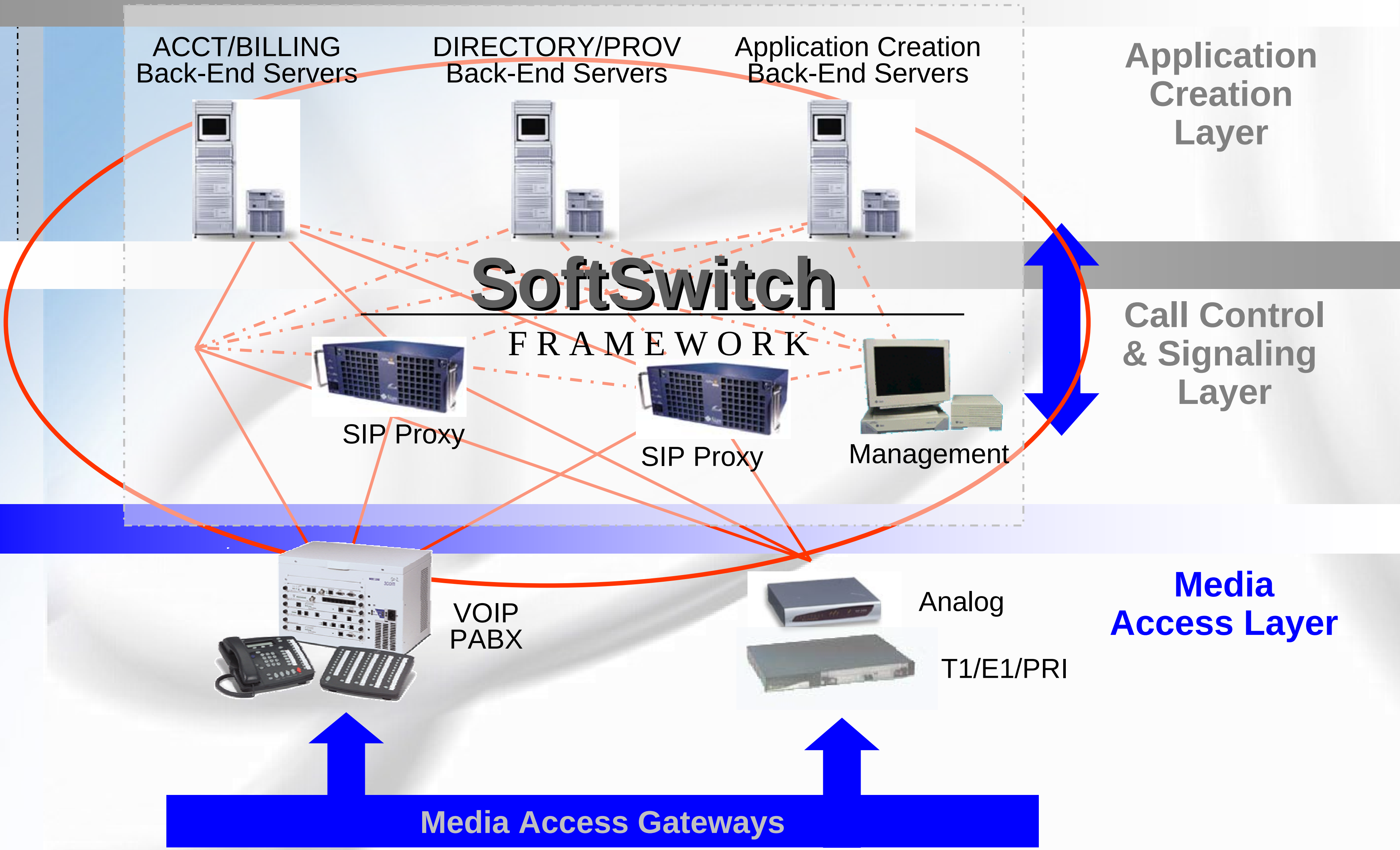



NBX 100 Communications System

SuperStack 3 NBX
Networked Telephony Solution

# 3Com Enterprise SoftSwitch

ARCHITECTURE



# Soluções 3Com Telefonia IP...
# *PABX em REDE*

**3Com NBX 100** → **Até 200 portas de Voz**

Pequenas e Médias Empresas
Escritórios Remotos

**3Com SuperStack 3 NBX** → **250 a 1500 portas de Voz**

Médias e Grandes Empresas
Multi-Site

# Soluçao Softswitch 3Com - VCX

**<u>Server Platform</u>** - Sun Fire V100 (supports up to 1500 calls)

Sun Fire V240 (supports up to 10,000 calls)

Sun Fire V880 (supports up to 50,000 calls)

# VCX Gateways Corpotativos

**<u>Analog Media Gateway</u>** (supports fax, modem, analog phones, analog trunks to PSTN) - 2, 4, 8, & 24-port

**<u>Digital Media Gateway</u>** (T1/E1/PRI connection to PBX or to PSTN) - 1,2,4,8, and 16 span.

Supports QSIG basic call, ANSI NI-2, ETSI Euro ISDN,CAS, and others.

**<u>PBX Media Gateway</u>** (station line connection to legacy PBXs provides VoIP integration) - 8 ports

# VCX – User Devices

**<u>IP Phones</u>** - NBX with dual loader code

3rd Party Phones - as needed

3Com Network Jack - Emergency Services (auto-location), Security

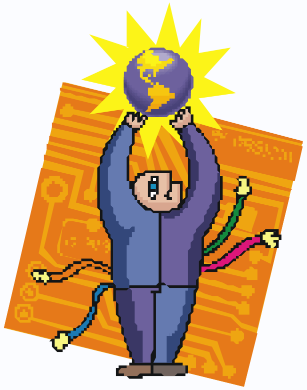
# VoIP Table Stakes

*(Supported by 3Com)*

- ✓ Transport and Trunking
- ✓ Call Forward
- ✓ Call Block
- ✓ Caller ID
- ✓ VPN Numbering Plans
- ✓ Call/Station Restriction
- ✓ Call Transfer
- ✓ Call Hold
- ✓ Call Waiting
- ✓ Call Park/Pickup
- ✓ Conference Calling
- ✓ Hunt & Trunk Groups
- ✓ Least Cost Routing
- ✓ Web Based Provisioning
- ✓ Toll Free
- ✓ 911/411 routing
- ✓ CDR Extraction & Billing
- ✓ Authentication/Rating
- ✓ Security
- ✓ Music on Hold

# Extensive Applications

- 3Com's powerful, proven suite of IP applications
- Easily managed centrally
- Deliver individualized, job requirement-based deployment throughout the enterprise
- Improving productivity:
  - ✓ **Voice Mail**
  - ✓ **Interactive Voice Response (IVR)**
  - ✓ **Unified Messaging**
  - ✓ **User Mobility**
  - ✓ **Personalized Call Treatment**
  - ✓ **Find Me/Follow Me Services**



Why 3Com?

# Resiliency

- VCX technology delivers five-nines availability
- Over 20 billion minutes served!
  - Customers include AT&T, one of the world's most demanding and highest volume SPs
  - Had a phone call recently with anyone in Florida, Texas, Colorado or Pennsylvania? Had a mobile call with anyone in Bejing or Shanghai? You've probably experienced reliable 3Com IP telephony.
- Reliability is a key 3Com commitment
  - NBX Rated 'excellent latency and voice quality metrics' in 2003 by Miercom
- 3Com's complete end-to-end high availability and security solutions to suit every business need