Scientists from all over the world need to collaborate with CERN on a daily basis. To improve this collaboration, the traditional telephone based audio-conference system needed to be modernized in three ways. Firstly, to provide the participants with more autonomy in the organization of their conferences; secondly, to eliminate the constraints of manual intervention by operators; and thirdly, to integrate the audio-conferences into a collaborative working framework.

The large number, and hence cost, of the conferences prohibited externalization and so the CERN telecommunications team drew up a specification to implement a new system. It was decided to use a new commercial collaborative conference solution based on the SIP protocol. The system was tested as the first European pilot and several improvements (as billing, security, redundancy...) were implemented based on CERN’s recommendations.

Audio-conference system evolution

Automatic audio-conference system evolution

- Traditional systems via PBX
  - Only available during working hours (0:00-18:00)
  - Limited to 25 participants per conference
  - Maximum of 15 simultaneous conferences
  - No cash for the organizer to manage conferences
  - No possibility to make outgoing calls

- Commercial solution adapted to our requirements
  - First prototype pilot
  - Available 24 hours a day from anywhere
  - Web-based application for managing conferences
  - No need to install software
  - No account required to participants
  - New features:
    - Call control (mute, hold, dross, lock...)
    - Instant messaging
    - File exchanging and download
    - Participant call-back possible
  - Voice communications:
    - Handle by PBX
    - MCU on the server
    - PBX/server connection via ISDN

Automatic audio-conference server (First implementation – 2005/2006)

- Fully redundant solution
- Call control features via SIP
- New features via web:
  - Application sharing
  - Online presentation
- Voice communications:
  - Handle by PBX
  - MCU on the server
  - PBX/server connection via ISDN


- Limit implementation of redundancy for this application worldwide
- Fully redundant solution
- Call control features via SIP
- New features via web:
  - Application sharing
  - Online presentation
- Voice communications:
  - Handle by PBX
  - MCU on the server
  - PBX/server connection via ISDN

Automatic audio-conference server operation

Audio-conference statistics

- Number of audio-conferences per year
- Daily calls distribution
- Cumulated hours per year

- Switchboard on the limit
- Capacity largely increased
- No limit for simultaneous conferences
- Same conference profile but 24h/7
- Conference management flexibility
- 65000 calls in 2008
- Large volume (50000 h/year)
- 400 simultaneous calls supported
- Automatic and scalable system

Audio-conference access

- Scheduled and ad-hoc conferences
- 400 ports for simultaneous calls
- URL to reach the conference web
- Possibility to call-back via the URL
- Simple access by phone (unique conference code)

Easy to use

- Web-based application for managing conferences
- No need to install software
- Leader organizes and fully controls his/her conferences
- Accessible 24h from anywhere
- E-mail / calendar invitations

Collaboration

- Instant messaging
- Application sharing
- Online presentation
- Conference recording and playback
- Detailed conference history

System connection

- Fully integrated with our PBX infrastructure
- Fully redundant system
- IP based solution
- Easy to maintain (no client software)
- Direct SIP calls being studied