CERN automatic audio-conference service

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Abstract. Scientists from all over the world need to collaborate with CERN on a daily basis. They must be able to communicate effectively on their joint projects at any time; as a result telephone conferences have become indispensable and widely used. Managed by 6 operators, CERN already has more than 20000 hours and 5700 audio-conferences per year. However, 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 audio-conference solution based on the SIP protocol.

The system was tested as the first European pilot and several improvements (such as billing, security, redundancy...) were implemented based on CERN’s recommendations.

The new automatic conference system has been operational since the second half of 2006. It is very popular for the users and has doubled the number of conferences in the past two years.

1. Introduction

CERN, the European Organization for Nuclear Research, is one of the world’s largest and most respected centres for scientific research. Its area of research is fundamental physics to discover the fundamental building blocks of the Universe and what forces act upon them. At CERN, the world’s largest and most complex scientific instruments are used to study the basic constituents of matter — the fundamental particles. By studying what happens when these particles collide, physicists learn more about the laws of Nature.

Founded in 1954, the CERN Laboratory sits astride the Franco–Swiss border near Geneva. It was one of Europe’s first joint ventures and now has 20 Member States. Currently, CERN employs around 2500 people, but some 8000 visiting scientists, half of the world’s particle physicists, come to CERN for their research. They represent 580 universities and 85 nationalities [1].

1.1. Audio-conference service

Having such a wide spread community of users, audio-conferences have become indispensable and widely used to communicate effectively on common projects. The traditional audio-conference system, run manually by 6 operators, already handled more than 5700 conferences and 20000 hours of communication every year. Nevertheless, the number of conferences was foreseen to grow drastically with the LHC start-up and the system itself needed to be modernized to improve the collaboration between scientists.
The traditional system had several constraints:
- Available only during working hours (8:00-18:00 CEST).
- Didn’t offer any tool for the organizer to manage conferences (managed by the operators).
- Limited to 29 participants per conference and to 15 simultaneous conferences.
- Didn’t offer the possibility to be called back by the system.

To address these issues, the new system needed to:
- Provide participants autonomy in the organization of their conferences, regardless of their geographical location and working environment.
- Avoid the constraints of manual intervention by operators.
- Integrate the audio-conferences into a collaborative working framework.
- Develop the system efficiency, increasing the number of simultaneous conferences and participants per conference.

2. Automatic audio-conference service implementation

The large number, and hence cost, of the conferences prohibited externalization, so the CERN telecommunications team drew up a specification to implement a new system.

The CERN strategy regarding telephony has been heading towards IP solutions for several years, so after evaluating different possibilities, it was decided to use a new commercial collaborative solution based on the SIP (Session Initiation Protocol) protocol [2]. It was subsequently tested as the first European pilot and adapted to our requirements and environment.

2.1. First implementation

During 2005, the solution was integrated into our PBX (Private Branch eXchange) via redundant ISDN (Integrated Services Digital Network) connections and tested both technically and functionally by the telecommunication team and a group of 50 users. Several improvements concerning security and operation were made and additional functionalities, such as integration into our billing system or resource management, were implemented.

The results of the pilot, the responsiveness of our supplier and integrator as well as the suitability of the solution to our requirements were all positive. Therefore, during the second half of 2006 the audio-conference service was put into production.

At this point, we had reached some of our goals:
- The new system was a web-based application (working environment independent) accessible 24 hours a day.
- Users could manage their conferences without installing software and avoiding manual intervention by the operators.
- Participants could access the conference without having an account and could be called back by the system.
- New functionalities, such as call control, instant messaging or call recording were introduced to go towards a collaborative tool.

Nevertheless, there were still some points to be addressed:
- The connection between the PBX and the system was based on ISDN although the solution was SIP-based.
- The system was handled by a single server and one single node of our PBX. In case either of these would suffer failure, the complete system would stop working.

2.2. Redundant solution

At the same time we were planning an upgrade of our PBX to make it fully IP; we contacted our integrator and supplier to investigate these two points. During 2007, different redundancy solutions were studied and discussed; the first implementation of redundancy for this application worldwide.
In the next figure, we can see the final scenario, where two audio-conference servers and two different PBXs are fully connected via SIP. Users employ the web application for collaboration purposes, call control and conference management but the PSTN (Public Switched Telephone Network) network for the voice:

![Image of audio-conference service operation](image)

**Figure 1: Automatic audio-conference service operation**

During 2007/2008, new collaborative features, such as online presentation and application sharing were proposed. Furthermore, the possibility of making direct SIP calls to the server so users don’t need to use the PSTN network is under study.

### 2.3. Features

From a functionality point of view, we can summarise the system as follows:

- **User interface**
  - Web-based application for managing conferences
  - No need to install software
  - Leader can organize and fully control his/her conferences
  - Accessible 24h from anywhere
  - E-mail / Calendar invitations

- **Conference access**
  - Scheduled and ad-hoc conferences (booking encouraged for capacity control)
  - 400 ports for simultaneous calls (no simultaneous conferences limit)
  - URL to reach the conference web.
  - Possibility to call-back via the URL
  - Simple access by phone (unique conference code)

- **Collaboration**
  - Instant messaging
  - Application sharing
  - Online presentation
  - Conference recording and playback
  - Detailed conference history
System connection
- Fully integrated into our PBX infrastructure
- Fully redundant system
- SIP based solution
- Easy to maintain

3. Results
The system has been operational since the second half of 2006 and is very popular for the users. The number of conferences has doubled in the past two years since its introduction.

As we can see in the figure 2, the old traditional audio-conference system was very much at its limit, handling around 5000 conferences and 20000 cumulated hours per year. The new automatic service has largely increase the capacity, counting more than 10000 audio-conferences, 50000 cumulated conference hours and 65000 participants in 2008 between both systems. Furthermore, there is no limit for simultaneous conferences or calls per conference. Currently, the new service supports 400 simultaneous calls to be shared between all the conferences, but it can be easily upgraded if needed.

The next figure shows the distribution of participants for both systems:
We can see the same pattern for both, the traditional and the new systems. However, the automatic service gives freedom and flexibility to the participants so they can book conferences 24h/7 independently of their time zone, which was one of our major requirements.

4. Conclusions
The CERN automatic audio-conference system received very positive feedback for the users due to its easiness to use, its flexibility for organizers, its conference control capacities and its collaboration tools.

From a technical point of view, it’s a solution that is easy to maintain, scalable, compatible with our telephony system and integrated into our working framework. Furthermore, it fits the CERN strategy to migrate the telephony towards IP solutions and to integrate it with advanced applications to offer more than simple telephony.

Financially it has also been interesting since it has reduced the human resources required to manage the audio-conferences. Moreover, including the call-back option, external or remote users can benefit from CERN’s attractive telephone tariffs. The next step, still under study, regards the connection via SIP to the conference service also for the voice, reducing the costs involved in telephone calls.

References
[1] CERN (www.cern.ch)
[2] Wallingford T 2005 Switching to VoIP (Sebastopol: O'Reilly)