Improved Communication through VoIP / Chat for a distributed pair programming tool (Saros)
Olaf Loga
Institut für Informatik
FU Berlin
14.01.10
Overview

• Motivation
• Requirements
• Audio Codec comparison (Speex & iLBC)
• Use Cases
• My Implementation
• Timeline
Motivation

• Easy communication in Saros without external applications

• Everyone who is in a shared project in Saros should have the possibility to communicate with each other member of the shared project (Chat & VoIP)

• Some logging options (e.g. the time someone is speaking / chatting, percentage of Saros sessions in which VoIP or Chat was used)

• Possibility to record the VoIP / chat conversation by the members of the session
Requirements (VoIP)

- VoIP Communication with more than 2 people
- One VoIP session in one shared project
- Everyone in a VoIP Session can invite everyone who is in the shared project
- The user who wants to use VoIP can adjust the codec settings (lower bit rate with IBB transmission)
- IBB, Jingle detection to automatically adjust the codec settings
Requirements (Chat)

- Chat with more than 2 people (Multi User Chat)
- One MultiUserChat (MUC) in one shared project in a separate Chat View
- Everyone in a shared project can join the MUC
- Problem of the actual chat implementation is that everyone who is using Saros is chatting with everyone else in one chatroom
Audio codec comparison

- Speex
- Supports bit rates from 2 kbit/s up to 44 kbit/s
- Supports variable bit rate encoding (VBR)
- Voice activity detection (VAD) (in combination with VBR encoding)
- Supports discontinuous transmission (in addition to VBR and VAD) → the transmission will be stopped when the background noise is stationary
- For example Speex is used in Asterisk, Mumble or Teamspeak
Audio codec comparison

- iLBC (internet low bitrate codec)
- Fixed bitrate (15.2 kbit/s – 20ms frames or 13.33 kbit/s – 30ms frames)
- No VBR support
- High robustness to packet loss

- For example iLBC is used in Skype, Google Talk, Yahoo Messenger or Gizmo5 (SIPphone)
Use Cases

Inviting s.o. to a VoIP Session:

**Scope**: A new invited user has joined a VoIP Session

**Actors**: Initiator, another User

**Precondition**:  
Both clients are in one shared project  
Both have correctly installed and configured record and playback devices (Headset)  
Record and playback device need to be set as default device in system sound options

**Success guarantee**:  
The user has been invited to the VoIP Session
Use Cases

Success scenario:
1. Initiator changes to the „Shared Project Session“ view
2. Initiator selects the user he wants to invite to a VoIP session. Afterwards he clicks on the telephone button in the „Shared Project Session“ view toolbar.
3. The selected user gets a yes-no popup.
4. User accepts the invitation
5. A new VoIP session will be started or the new user will be added to an existing VoIP session.

If the user doesn’t accept the invitation the initiator will be informed that his invitation has been rejected.
Use Cases

Leave a VoIP Session:

**Scope:** One Client leaves the VoIP Session

**Actors:** Client

**Precondition:**
- The client is in a VoIP Session

**Success scenario:**
- Client changes to the „Shared Project Session“ view
- Client clicks on the „telephone_stop“ button
- Client leaves the VoIP Session
- (Other members of the VoIP Session will be informed)
My Implementation

- VoIP in Saros uses jSpeex (Java Speex API) as Encoder / Decoder (at the moment)

  Problem: The audio playback is stuttering

- VoIP uses the StreamService (by Stephan Lau)
- The VoIP function doesn’t use the XMPP VoIP Interface because of some problems with identification of capture and playback devices. (tested with the Spark IM)
- Existing chat function is using the XMPP Extension for Multi User Chat (XEP-0045)
Timeline

- **Week 0 (07.12 – 11.12):**
  - Introduction to Saros
  - Introduction to XMPP
  - Introduction to Speex Java API (jspeex)
  - Adding already implemented Chat to Saros and testing

- **Week 1 (14.12 – 18.12):**
  - First prototype for audio encoding (record sth. and play it) with JSpeex

- **Week 2 & 3 (20.12 – 08.01):**
  - Simple integration in Saros (with some problems, e.g. stuttering audio playback)
  - Registration of the Bachelor thesis (official deadline: 01.04)

- **Week 4 (11.01 – 15.01):**
  - Fixing the stuttering audio playback
  - Preparation for the introduction presentation

- **Week 5 & 6 (18.01 – 29.01):**
  - Testing and adding other codecs
  - VoIP Session with more than 2 persons
  - User Interface
Timeline

• **Week 7 & 8 (01.02 – 13.02)**
  - Adding logging functionality
  - Revision of the Chat functionality
  - Testing
  - Error Handling
  - Preparing for Release

• **Week 9 (15.02 – 19.02)**
  - Probably Release week

• **Week 10 & 11 (22.02 – 05.03)**
  - Maybe fixing some bugs
  - Starting written thesis

• **Week 12 (15.03 – 19.03)**
  - Finishing written thesis
Sources

- http://fmj-sf.net/
- http://www.ilbcfreeware.org/
- http://speex.org/
- http://jspeex.sourceforge.net/
- http://www.voip-info.org/
Vielen Dank!