Disruption tolerance for SIP

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Outline

- Objective of this thesis
- Prototype
- Use case
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- Experimental SIP network
- Test cases
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- Conclusion

Objective of this thesis

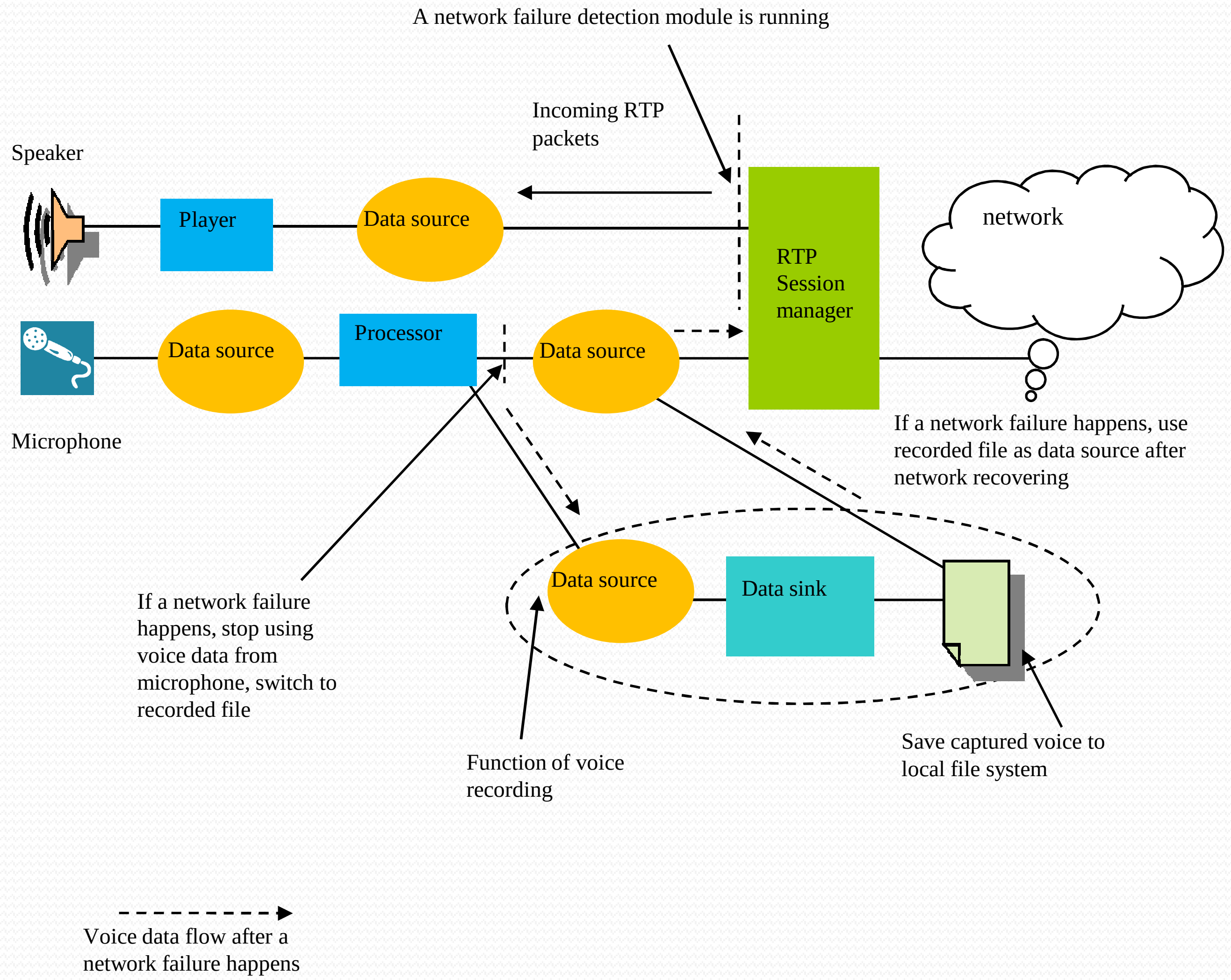
- Wireless networks are unstable
- Ubiquitous wireless network coverage is difficult to deploy
- Users may lose connection temporarily when travel through different wireless networks
- This thesis provides an alternative solution of disconnection tolerance, services as a complementary solution to improve usability

Prototype

- The prototype is implemented in Java
- Builds upon open source project – SIP Communicator
- Uses JMF (Java Multimedia Framework) as development library
- Media path based disconnection detection mechanism
- Recovers the broken call automatically and replays the unheard voice
- Uses voice mail server as permanent repository

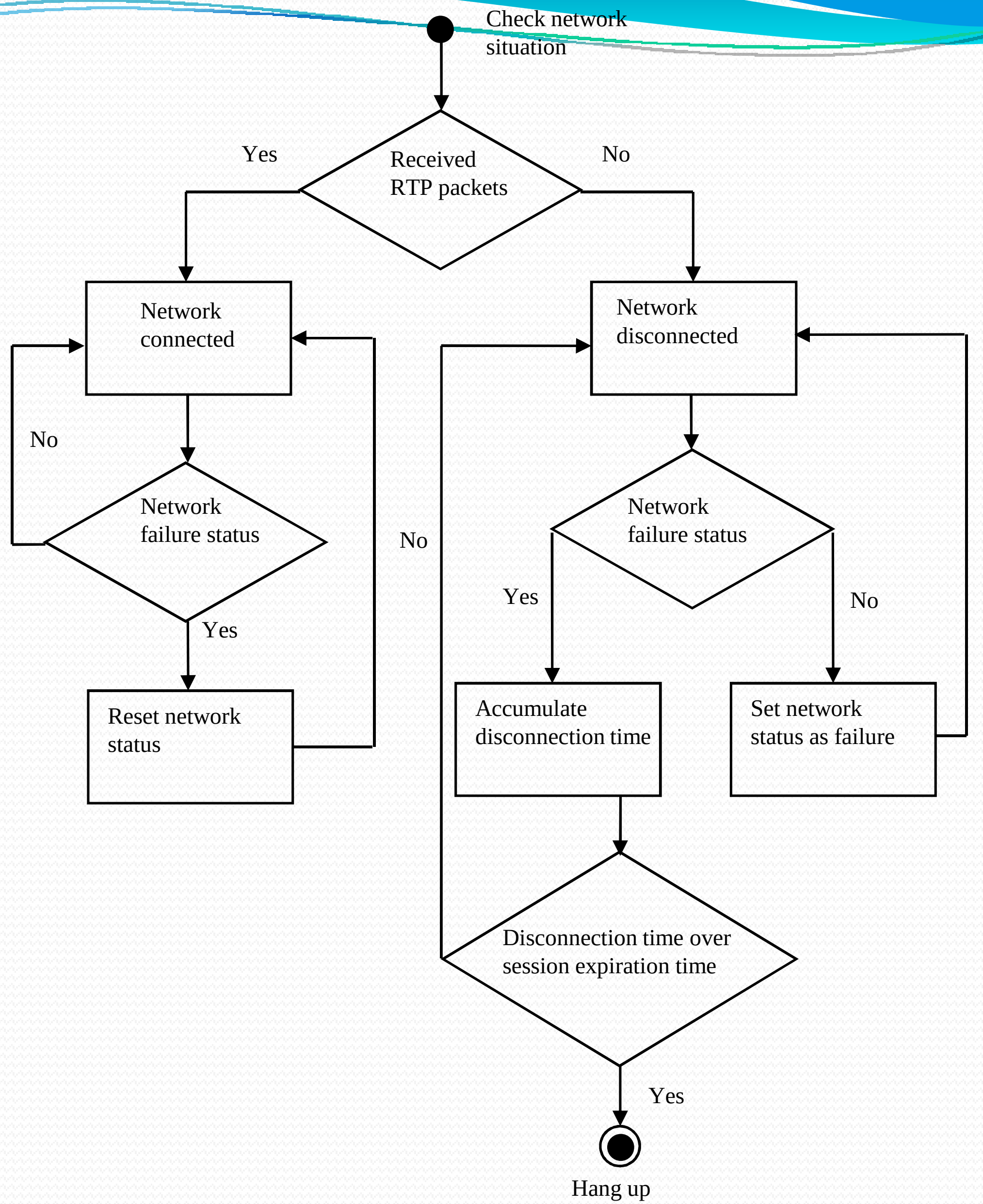
Use cases

- Short duration of network failure
  - Recover the broken call by restoring the RTP session
  - Replay the saved voice
- Long duration of network failure
  - Recover the broken call by re-INVITE
  - Replay the saved voice
- Unpredictable disconnection
  - Connect to the voice mail server and save the unheard voice data
  - User can fetch the saved voice data later

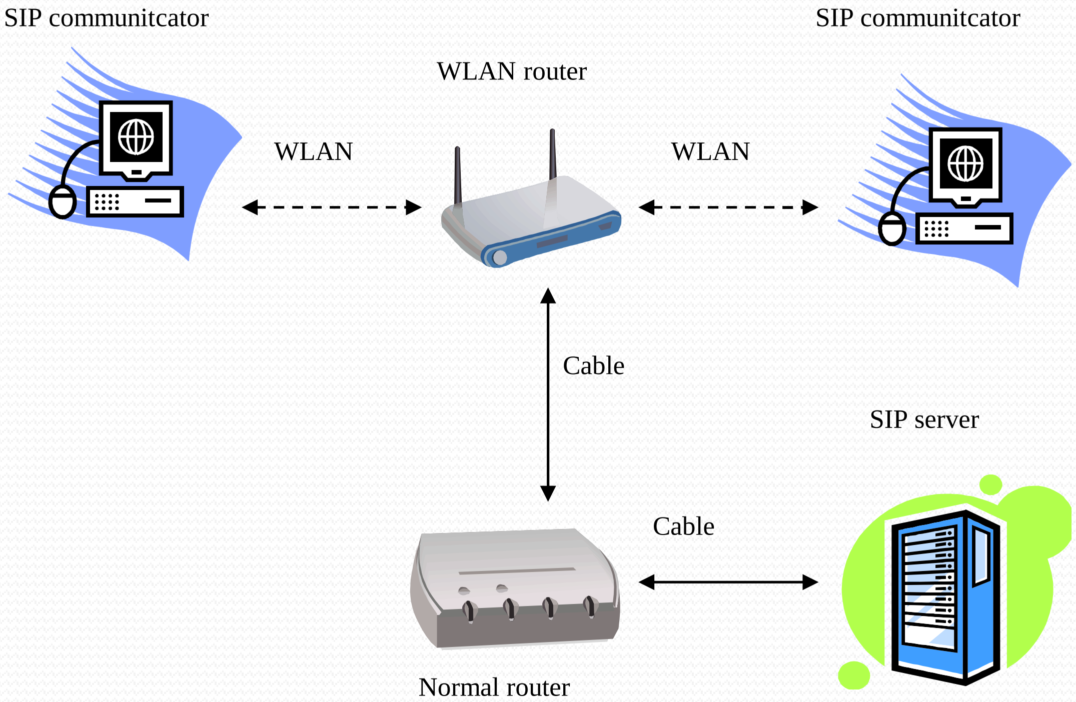

A network failure detection module is running
Incoming RTP packets
Speaker
Player
Data source
RTP Session manager
network
Data source
Processor
Data source
Microphone
If a network failure happens, use recorded file as data source after network recovering
Data source
Data sink
If a network failure happens, stop using voice data from microphone, switch to recorded file
Function of voice recording
Save captured voice to local file system
Voice data flow after a network failure happens

Modules

- Sound record module
  - Record voice data to local file system
  - Multiple temporary files are used to ensure read/write at the same time
- Network failure detection module
  - RTP/RTCP packets based detection mechanism
  - Doesn’t require to understand the low level network interfaces
  - Imprecise detection


Check network
situation
Received
RTP packets
Yes
No
Network
connected
Network
disconnected
No
Network
failure status
Yes
Reset network
status
No
Network
failure status
Yes
No
Accumulate
disconnection time
Set network
status as failure
Disconnection time over
session expiration time
Yes
Hang up

Experimental SIP network



Test cases

- Normal call
  - Works fine
- Short duration of network failure
  - Works as assumed, the recorded voice comes with a large delay after the call is recovered
- Long duration of network failure
  - Works as assumed, the broken call is rebuilt
- Unpredictable network disconnection
  - Connects to the voice mail server but the user can't interact with the voice mail server

Enhancements

- Limited SIP messages are implemented in the SIP Communicator
- DTMF function is missing
- Silence suppression is not implemented in this thesis
- No user interface to switch off the disruption tolerance function

Conclusion

- Dealing with disruption tolerance mechanism for SIP based real-time communication
- Most use cases are working as assumed
- Future work is needed to improve both usability and stability

Thank you!

Questions?